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Alternative Methods of Measuring Acoustic Absorption

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ABSTRACT OF THE
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In order for room acoustics to be planned, the most crucial information required is how sound waves are affected as they hit the boundaries of a given room. As a sound wave hits a material, parts of the wave reflect back into the room, transmit outside of the room and absorb into the material. The most important information of a material from an acoustic engineer's point of view is the acoustic absorption.

Three standardized methods exist to measure the acoustic absorption of a given material. The objective of this thesis was to explore two non-standardized methods and to determine whether reliable information could be gathered using these alternative methods. The alternative methods consist of a reverberation room and a scale model measurement.

The alternative methods were used to measure the acoustic absorption of a glasswool board, Ecophon Master C. The results were then compared to existing laboratory results provided by the manufacturer of the board. This thesis also reports impedance tube measurements and omnidirectionality testing of two cube shaped loudspeakers. The results showed that the reverberation room method outperformed the two other methods in terms of reliability and correlation with the reference results. In conclusion, the reverberation room method is the only method that produces sufficiently reliable results to compare the room acoustic behaviour of different materials.

Cube shaped loudspeakers cannot be exclusively be concluded to be omnidirectional, although if designed correctly, it is possible to create a cube shaped omnidirectional loudspeaker. Due to uncertainties in reverberation time measurements, it is known that inter-laboratory results are not good. ISO currently has a work group trying to make the standardized reverberation chamber method more uniform between laboratories.

Keywords: Acoustic impedance, acoustic measurements, acoustic radiators, acoustic signal analysis, acoustics, architectural acoustics, standing wave measurements

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Jotta tilan akustiikkaa olisi mahdollista suunnitella etukäteen, on tiedettävä kuinka ääniaallot käyttäytyvät osuessaan materiaalien pintoihin. Kun ääniaalto osuu materiaaliin, osia aallosta heijastuu, läpäisee ja absorboituu materiaaliin. Absorptio on akustisesta näkökulmasta materiaalin tärkein ominaisuus.

Akustista absorptiota on mahdollista mitata kolmella standardoidulla menetelmällä. Tämän diplomityön tarkoituksena oli tutkia, voiko kahta standardoimatonta menetelmää käyttämällä saada luotettavia tuloksia. Tutkitut vaihtoehtoiset menetelmät olivat mittaus tyhjässä huoneessa ja pienoismallimittaus.

Vaihtoehtoisilla menetelmillä mitattiin Ecophon Master C - akustiikkalevyä. Tuloksia verrattin valmistajilta saatuihin standardoiduilla laboratoriomittauksilla tuotettuihin tuloksiin. Tässä diplomityössä esitetään myös impedanssiputkimittaus ja kahden kuution muotoisen kaiuttimen ympärisäteilevyysmittaukset.

Tulokset osoittivat mittauksen tyhjässä huoneessa ylivertaiseksi muihin tarkasteltuihin metodeihin nähden. Kyseinen menetelmä on ainoa, jolla materiaaleja vertaava mittaaminen on mahdollista. Kuution muotoisia kaiuttimia ei voida mittausten perusteella sanoa eksplisiittisesti ympärisäteileviksi, mutta kuution muotoinen ympärisäteilevä kaiutin on mahdollista valmistaa.

Johtuen jälkikaiunta-ajan mittauksen epävarmuustekijöistä, on tiedossa, ettei laboratorioiden väliset tulokset ole kovin yhtenäisiä. ISO:lla on tällä hetkellä työryhmä, jonka tehtävänä on tutkia, miten standardoitua kaiuntahuonemenetelmää saisi myös laboratorioiden kesken yhtenäisemmäksi.

Avainsanat: Akustinen impedanssi, akustiset mittaukset, akustiset säteilijät, akustinen signaalianalyysi, akustiikka, rakennusakustiikka, seisovan aallon mittaukset

Preface

This Master's thesis was done in and funded by Akukon Consulting Engineers Ltd. The work took place between December 2010 and May 2011

I want to thank my instructor Mr. Henrik Möller. He has been very supportive and helped greatly on practical matters in the acoustic measurements performed for this thesis. He has also been a great help when clarifying the research question at the initial phases of the work.

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I would also like to thank all my co-workers and friends who have been supportive throughout the course of this work.

Finally, my deepest gratitude goes to my beloved Elli, who not only supported me throughout this work, but also took time to translate my colloquial Finglish to proper English.

Otaniemi, 23.5.2011

Joonas Jaatinen

Contents

Symbols and abbreviations

Symbols

c	Speed	T	Temperature
f	Frequency	λ	Wavelength
I	Intensity	m	Sound attenuation coefficient
L_p	Sound pressure level	L_i	Sound intensity level
p	Pressure	$\mathbf{R}$	Reflection factor
R	Amplitude of a reflected wave	$\mathbf{Z}$	Acoustic impedance
v_n	Particle velocity	S	Wall area
α	Absorption coefficient	T_{60}	Reverberation time
V	Volume	A	Absorption area (Sabine)
A'	Absorption area (Eyring)	L_{xyz}	Length of room dimension
$x(t)$	Signal	$f(t)$	Filter
δ	Unit-impulse	$h(t)$	Impulse response
$\mathbf{P}$	Sound power	s	Standing wave ratio
$\mathbf{H}_{12}$	Complex transfer function	γ	Phase
$\mathbf{P}_s$	Power emitted by a sound source	r	Reflection factor
u	Particle velocity	k	Wave number
d	Distance	l	Length
ϵ_{20}	Spatial variance of sound field		

Abbreviations

dB	Decibel
FFT	Fast fourier transform
Hz	Hertz
ISO	International organization for standardization
IT	Impedance tube
MLS	Maximum length sequence
RR	Reverberation room
SNR	Signal-to-noise ratio
SPL	Sound pressure level

1 Introduction

In order for room acoustics to be planned, the most crucial information required is how sound waves are affected as they hit the walls, the ceiling, the floor and the furniture of a given room. Boundaries, size, shape and objects inside a room are the factors that make a space sound the way it does.

The sound field of a room is described as a system wherein a vibrating sound source causes a disturbance, which then disperses into the room as a sound field through the medium [17]. As sound propagates inside the room it hits the room boundaries reflecting it back into the room, absorbs into the boundary materials and transmits outside of the room.

How the boundaries affect a sound field is mostly a combination of the aforementioned phenomena; reflection, absorption and transmission. These characteristics fully describe a construction's or a single material's acoustic behavior in a room [17]. The most important information of a material from an acoustic engineer's point of view is acoustic absorption. Materials are usually categorized as absorptive or reflective. Measurement methods have been developed and standardized in order to enable the study of the acoustic properties of different materials.

There are three standardized methods of measuring acoustic absorption; the reverberation chamber method described in the standard ISO-354 [3] and two impedance tube measurements that are described in standards ISO-10534-1 and ISO-10534-2 [8] [9]. The measurements performed using an impedance tube are accurate for normal incident sound waves only [8] [9]. Thus, the reverberation chamber measurement is the only standardized method for measuring acoustic absorption for all sound incident angles [3].

A reverberation chamber is a reverberant room with a volume of approximately 200 m^3. The chamber has some restrictions in room geometry and background noise, these will be discussed in further detail in Section 3.2. Reverberation chambers are not widely available in southern Finland, which has proposed some challenges for acoustic consultants and product developers.

The objective of this thesis is to analyze and study data aquired using alternative methods of measuring acoustic absorption along with data from measurements carried out in an impedance tube using the transfer function method. Both of these studies will be carried out on identical material. In conclusion, the results will be compared with existing reverberation chamber measurement data.

1.1 Research problem statement

Three standardized methods exist to measure the acoustic absorption of a given material. Only one of these is universally recognized as an acceptable method to provide data that can be used to officially report the acoustic absorption of a material. The objective of this thesis is to explore two non-standardized methods and to determine whether reliable information can be gathered using these alternative methods. These studies will be carried out on one material that has been previously measured in a reverberation chamber according to the ISO-354 standard [3].

2 Theory

In this section, sound theory and necessary topics in room acoustic theory is discussed.

2.1 Sound Theory

2.1.1 Sound Propagation

Sound is an oscillating mechanical wave that transmits through a medium. A medium can be either a solid, liquid, or gas. A sound wave propagates through a medium with a speed of c. A common medium for a sound wave is air. The speed of a sound wave propagating through air can be calculated using the following equation [3]

$$c(T) = (331.3 + 0.6T/^{0}C)\left[\frac{\mathrm{m}}{\mathrm{s}}\right], \tag{1}$$

where T is temperature.

Sound waves cause vibrations in the particles of the medium, which may not necessarily always be in the same phase. These vibrations are caused by the nature of an oscillating sound source, for example, when a loudspeaker diaphragm moves forward it causes the gas to compress and as it changes direction it causes a rarefaction of the gas near it. Thus, it is common to find points in a sound field where particles are distributed more sparsely or densely [17].

The density of the medium determines the ease, distance and speed of sound transmission. The higher the density of the medium, the slower sound travels through it [17].

Sound waves have unique characteristics, they vary in frequency, loudness and quality. A sound wave is said to have a high or low pitch depending on its frequency and in turn frequency is relative to the length of the sound wave. A short wavelength is heard as a high pitch and hence, has a high frequency and vice versa.

As shown below, sound frequency and wavelength are also dependant on the speed of the sound wave[17]:

$$f = \frac{c}{\lambda}[\frac{1}{s}, Hz] \tag{2}$$

Sound quality is an abstract categorization between *tones* and *noise*, or more commonly pleasant and unpleasant sound, respectively. Sound quality will not be discussed any further. Sound loudness, however will be described in the following chapters.

Sound propagation in a medium can be described as a system where a sound source causes vibrations in the particles of the medium. Sound waves attenuate while propagating in a medium. Attenuation in air is caused by two factors:

- Losses by heat conduction and viscosity in the air (m')
- Temperature dependant rotational and vibration relaxation of the molecules in air (m'').

Sound pressure is the physical quantity of particle displacement which is caused by the sound source on the medium [17]. Sound intensity is the product of the particle velocity and sound pressure. The attenuation of sound intensity in air is given by [18]

$$I(x) = I_0 e^{(-m'+m'')x}, \tag{3}$$

where I is intensity with respect to the distance of the source. The attenuation coefficient, m, is dependent on frequency, temperature and relative humidity. In practice, sound attenuation by medium becomes a significant source of error at high frequencies, especially when measuring material properties in large spaces.

Media, other than air, will not be discussed in this thesis. All the methods which will be studied will have air as a medium for sound wave traversal. Although the speed of sound waves can vary quite heavily for different gases, sound waves behave similarly in all gases.

2.1.2 Sound Levels and Human Hearing

Sound Pressure Level (SPL) is measured using a logarithmic decibel scale. The logarithmic nature of the decibel scale most naturally models how human ears percieve an acoustic stimulus. A decibel is a scalar unit that describes the ratio between two numbers, one of which is a reference point. For sound pressure, this reference point is 20 micropascals due to the fact that it is considered to be the threshold for human hearing at a frequency of 1 kHz. Sound pressure level is given by [17]

$$L_p = 20\log_{10}(\frac{p}{p_0})[\text{dB}], \tag{4}$$

where L_p is sound pressure level, p is the measured pressure level and p_0 is the reference value described above.

Sound intensity level is used to describe the flow of acoustic energy. Therefore, it cannot be measured using a traditional microphone, as traditional microphones only measure sound pressure, p. Sound pressure is therefore a sound field quantity whereas sound intensity is an energy quantity. In other words, Sound pressure is the effect caused by sound intensity [17]:

$$L_I = 10\log_{10}(\frac{I}{I_0})[\text{dB}], \tag{5}$$

where I_0 is the reference value for sound intensity, $10^{-12}W/m^2$.

Sound pressure is the measurable quantity that moves the human ear drum. It can be measured objectively using electroacoustic devices and sound intensity can be calculated theoretically. Sound intensity microphones have been a subject of development and although sound intensity microphones have recently been created, this thesis will not cover such measurements.

Although values of SPL are referenced to human hearing, SPL does not subjectively indicate how loudly humans hear a tone as human hearing is dependant on frequency and amplitude. A scaling and weighting system for SPL exists, which takes human hearing into consideration. However, these are not relevant considering that SPL is not measured, only percieved using human ears.

2.1.3 Sound Absorption

In contrast to sound propagation being unbound and attenuated only by its medium, room acoustics refers to sound propagating in a bounded space. The boundaries usually reflect, absorb and transmit sound. The combination of the aforementioned phenomena is shown in Figure 1.

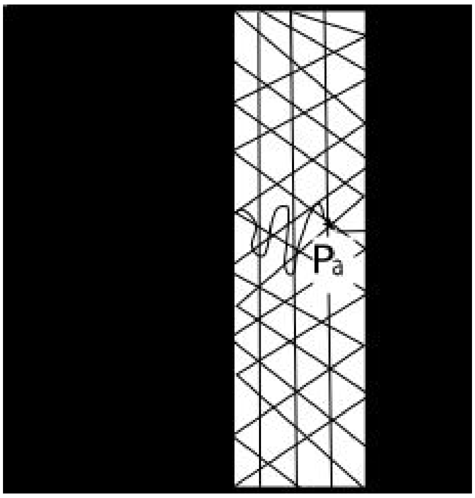

Figure 1: The behaviour of the incident sound wave as it hits a wall

When a sound wave hits a wall a fraction of that wave is reflected back and the wall becomes a new point of origin for the wave. Changes in amplitude and phase in comparison to the incident wave can be observed as a reflection factor [8]

$$\mathbf{R} = |R|e^{(i\gamma)}. \tag{6}$$

Note here, that R is a property of the wall, not the wave.

The amplitude of the reflected wave is a fraction of the incident wave. This fraction can be written as shown below and is called the absorption coefficient of the wall [8]:

$$\alpha = 1 - |R|^2. \tag{7}$$

The absorption coefficient α is frequency dependent and its values range between 0 and 1. If the coefficient is 1, the wall is said to be totally absorbent. In this case R has a value of 0, which means that the wall absorbs all of the sound power.

As for the other extreme, when the absorption coefficient has a value of 0, R can be either 1 or -1. When R is 1 the wall is said to be hard. In the case where R is -1 the wall is said to be soft. The latter case occurs rarely in room acoustics and if so, only on a limited frequency range [17].

From a room acoustics point of view, the reflection factor describes the wall properties for all frequencies and angles of incidence.

The absorption coefficient depends on the incident angle, so for a wall or structure it is common to use an average descriptor of absorption. Usually a wall is not uniformly absorbent but depends on the locations of different wall elements. If the wall has an area S which consists of parts S_1 and S_2 with absorption α_1 and α_2 respectively, then the total absorption of the wall is [17]

$$A = \frac{1}{S}(S_1\alpha_1 + S_2\alpha_2). \tag{8}$$

Equation 8 states that different absorption areas on a wall are calculated as a weighted sum of the whole wall.

Another quantity for describing wall properties is impedance, a factor which is related to the physical behaviour of the wall. Wall impedance is the ratio between sound pressure and particle velocity perpendicular to the wall [17]:

$$\mathbf{Z} = (\frac{p}{\bar{v}_n})_{surface}. \tag{9}$$

Like the reflection factor, wall impedance is also a complex value. Complex values describe acoustic properties: Sound pressure (real part) and phase (imaginary part). Wall impedance will be described further in chapter 3.

2.2 Room Acoustics

In earlier parts of this chapter, the properties of sound waves have been described. However, these descriptions do not take into consideration the effect of room boundaries on sound travelling within a room. The methods of analysis presented in this thesis require the use of bound spaces, therefore the theory of sound in an enclosed space must be reviewed.

2.2.1 Room Build-Up and Decay time

When a sound source starts emitting inside a room, the initial sound waves reflect from room boundary to boundary, gradually building up sound pressure in the room. Eventually, equilibrium is formed between the sound energy transmitted by the loudspeaker and the absorption of the wall. The time taken from when the sound source started emitting on to the time that equilibrium is achieved is called the build up time of a room.

Sound energy density is [17]

$$\mathbf{E} = \frac{I}{c}, \tag{10}$$

and the sound power transmitted by a source is P_s. The balance between absorbed energy and energy transmitted by the sound source can be written as [11]

$$P_s \mathrm{dt} - \frac{1}{4}\mathbf{E}cA \,\mathrm{d}t = V\frac{\mathrm{d}\mathbf{E}}{\mathrm{d}t}\,\mathrm{d}t. \tag{11}$$

When the equilibrium described earlier has been reached, $\frac{\mathrm{d}\mathbf{E}}{\mathrm{d}t} = 0$, and the stationary energy density can be written [11]:

$$\mathbf{E}_0 = \frac{4P_s}{cA}. \tag{12}$$

Equation 12 can also be written as [11]

$$\mathbf{E} = \mathbf{E}_0(1 - e^{\frac{cA}{4V}t}), \tag{13}$$

and when the energy density reaches its maximum and the sound source is switched of, Equation 11 can be written as [11]

$$V\frac{\mathrm{d}\mathbf{E}}{\mathrm{d}t} = \frac{1}{4}\mathbf{E}cA, \tag{14}$$

Which, when integrated gives [11]

$$\mathbf{E} = \mathbf{E}_0 e^{\frac{cA}{4V}t}, \tag{15}$$

this shows that the sound field inside a room both builds up and decays as an exponential function. Equation 15 can be therefore used to calculate a rooms' ideal build up and decay time.

2.2.2 Reverberation Time

As sound waves travel within room boundaries they lose energy due to absorption and transmission as well as the attenuation of air. The prediction of reverberation time begun with research by W.C Sabine in the beginning of the twentieth century. Sabine concluded a simple formula which was based on the ratio between room volume and the amount of absorbing material inside the room. The reverberation time is defined as the time it takes for a room's sound field to dampen 60 decibels after a sound source is turned off. In practice, this is the moment at which the sound becomes inaudible.

Sabine's equation can be derived from the room decay function described earlier. Sabine concluded that after the build-up time, a room's sound field decays 60 decibels [11]:

$$\ln(\frac{\mathbf{E}}{\mathbf{E}_0}) = \ln 10^{-6} = -\frac{cA}{4V}t. \tag{16}$$

When t is solved from the equation written above, the result is Sabine's equation.

Sabine's original equation was [16]

$$T_{60} = 0.161\frac{V}{A}[s], \tag{17}$$

However, this equation did not originally account for medium attenuation, and so it was later introduced as a correction term [6]:

$$T_{60} = 0.161\frac{V}{A + 4mV}[s], \tag{18}$$

where V is room volume in cubic metres, A is the total absorption as calculated in Equation 8 and m is the energy attenuation constant defined earlier in this chapter. With this equation, zero reverberation time requires infinite absorption. Sabine has reported that an open window, with no reflected sound, has an absorption coefficient of 1. However, Sabine has also reported absorption coefficients higher than 1, which according to equations 17 and 18 seems theoretically impossible. When the Sabine unit of absorption is used, absorption values greater than 1 are the result of neglecting the edge effect, for example.This effect states that objects placed on a wall tend to have more than one absorptive edge, as is the case with, sign boards or balconies in a concert hall, for example.

The reverberation time depends on the volume of the room and the rate at which the sound energy is absorbed by the boundaries and the objects in the room. Reverberation time is the least subjective of all the physical criteria used in rating room quality as it can be objectively measured. Note here that reverberation time in itself does not give any qualitative information on the acoustics of a space.

Predicting and calculating the reverberation time of a room is an ongoing challenge for acousticians. A formula or software to completely describe the reverberation time of a room regardless of room size, shape and absorption amount has yet to be discovered.

Eyring presented an alternate equation to Sabine's in 1930. This equation states that in a room where all boundaries are fully absorptive, reverberation time is zero. Eyrings equation is [27]

$$T_{60} = 0.161 \frac{V}{A' + 4mV}[s]. \tag{19}$$

As can be seen, the equation differs from Sabines equation only in the way the total absorption is calculated. Sabine calculates the weighted sum of the absorptive area, whereas Eyring calculates it as follows [27]

$$A' = S_{tot}[-2.30 \log_{10}(1 - \alpha_{ey})][m^2]. \tag{20}$$

The Sabine and Eyring equations require different assumptions. Sabine assumed that a sound wave travels by impacting the boundaries one by one while Eyring based his formula on the assumtion that all surfaces are simultaneously impacted by the initial sound wave [17]. Neither of these assumptions are true for all rooms.

2.2.3 Statistical Room Acoustics

Calculations for a sound field in a room could be carried out using the equations presented in Section 2.1.First the sound field emitted by the source must be calculated to which the effect of each reflection is then added until the sound field is built up to its maximum. After the initial reflections, for example the sound field reflected from the floor is reflected again from the ceiling and the walls and so on. Although this procedure would result in a complete description of the sound field, the amount of calculations required is so vast that even with todays calculation technology, it would be very time consuming to carry out. For simple room structures like a cubic room this method can be applied in a simplified form, however for larger and more complex rooms like a movie theater or a concert hall there are different approaches to compute a resulting sound field.

In practice, the sound field of any given room is complicated, which makes it difficult to model. Theoretic computational models for room acoustics do exist, however, they cannot be applied in practical situations as exact models because boundary conditions for irregular room shapes, nonrigid walls and various furniture are unclear. **Heinrich Kuttruff** states in his book that [17]

... "it is not very promising to apply the methods of wave theory in order to find answers to questions of practical interest, especially if the room under consideration is large and somewhat irregular in shape."

"Questions of practical interest" meaning, for example, prediction of reverberation time or calculating resonant frequencies in rooms that are more complex.

A sound field is dependant on the shape of the sound source [6] although, from a sufficiently long distances all sound sources can be considered to produce a spherical wave. An omnidirectional source results in a spherical wave. This leads to an important sound distance law, the sound pressure level of all spherical waves is inversely-proportional to the square of the distance from the source $(1/R^2)$.

For the purpose of practical analysis, from a sufficiently long distance from the sound source, the curvature of spherical waves can be neglected [17]. The spherical wave is thus considered as a plane wave, which simplifies calculations. This assumption causes a negligible error in calculations [17].

2.2.4 Geometrical Room Acoustics

For larger, more complex spaces room acoustics can be modelled using geometrical room acoustics. This concept models sound waves as sound rays that are emitted from a source and reflected from boundaries until they are completely attenuated [17]. The behaviour of these sound rays resemble the laws of optics but model the acoustic behaviour well enough in order to be used in practical applications such as in concert hall modeling. This method is also known as ray tracing and serves as a basis for most room-acoustics software.

2.2.5 Room Modes

Standing waves in a room are called room modes, these occur when the wavelength of a sound wave is an exact multiple of the room dimensions. The standing sound waves resonate between the room boundaries creating nodes and antinodes at different locations in the room. The lower the multiple of a room dimension the louder the resonance. A room mode can be thought of as a temporary storage of acoustic energy. Room modes are location specific, one can hear them by walking around a room where low frequency sounds are present, for example.

Room modes for a rectangular room can be calculated using the following formula [17]:

$$f_{n_x n_y n_z} = \frac{c}{2\pi}\left[\left(\frac{n_x}{L_x}\right)^2 + \left(\frac{n_y}{L_y}\right)^2 + \left(\frac{n_y}{L_y}\right)^2\right][\frac{1}{s}], \tag{21}$$

where L is the room dimension (length x, depth y and height z) and n is an integer multiple of a room dimension. For example, lengthwise, the frequency of the first mode in a room is calculated using $n_x = 1$, $n_y = 0$ and $n_z = 0$. Modes up to 200 Hz in a rectangular room of size 3 m $\times$ 4 m $\times$ 2.5 m are shown in Figure 2

As the multiples increase, room modes become more frequent and at higher frequencies room modes no longer dominate the acoustics of the room. Manfred Schroeder developed a formula for calculating the critical frequency over which modes are not a dominant part of a rooms acoustics [20]:

$$f_s = 2000\sqrt{\frac{T_{60}}{V}}. \tag{22}$$

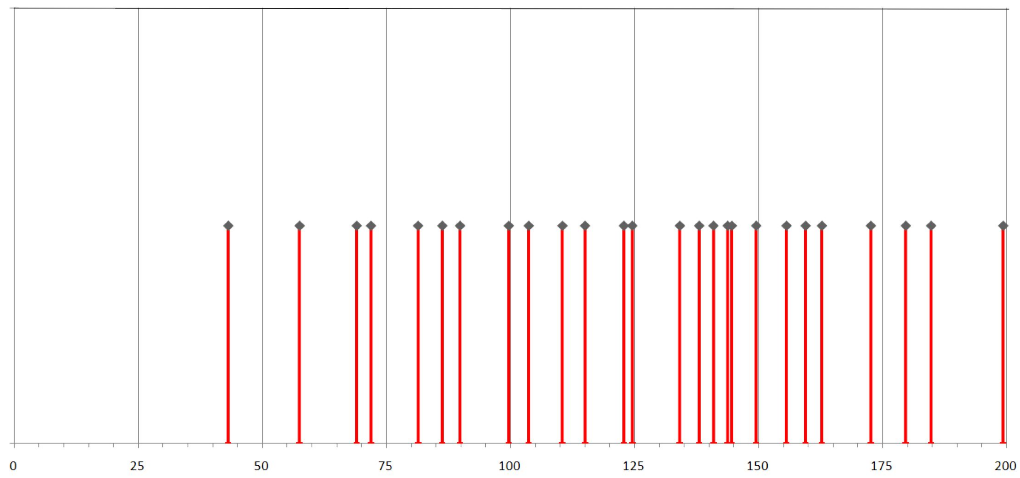

Figure 2: Modes in a rectangular room, calculated using Equation 21

The Schroeder-frequency usually has an effect on the design of small spaces such as studios, reverberation chambers and sound control rooms, but for larger spaces such as churches and concert halls, the critical frequency is so low that individual room modes are not a problem. Some of the measurements in this thesis will be performed in spaces where the Schroeder-frequency is over the typical measuring range. Results below the Schroeder-frequency should be considered with a certain amount of doubt. Room modes make for one of the limits for accurate measurements in smaller rooms.

3 Methods for Measuring Acoustic Absorption

3.1 Measuring Reverberation Time

Most methods of analysis regarding room acoustics, including absorption measurements, require the use of a reverberation time measurement. Reverberation time of a space can be measured using various types of equipment. The most common method at present uses a loudspeaker as a sound source, a microphone and a computer run software which controls and analyzes the measurement data.

The definition of reverberation time is quite simple, however actually physically measuring the T_{60}-parameter is not simply defined [6]. The task is to find how the room responds to an impulse. In traditional signal processing language, the room represents a system of which the impulse response is a property. In theory, a room is excited using a unit impulse from which the response is recorded inside of the room however in practice, a unit-impulse or the Dirac delta-pulse is impossible to create. Nonetheless, there are methods to measure the impulse response of a room with sufficient accuracy by using a sweep, noise or impulse like - signal. Measuring room impulse responses is the most common task for an acoustician. The reason for this, is that many room acoustic descriptors like the reverberation time, for example, can be derived from room impulse responses.

Sabine measured reverberation times subjectively using organ pipes to create sound pressure at a specific frequency and a stop-watch to measure the time from when he turned off the pipe to when he could no longer hear the sound of the pipe. Although measurement equipment has since since then modernized, Sabine's subjective method is quite similar to how room acoustics are measured today.

In reverberation time measurements, a sound source, a pressure sensor (microphone) and a recording device are required for analysis. Methods tend to vary, in sound source and measurement signal. All of the methods are aimed to produce an impulse response from which the value of the T_{60} parameter can be calculated from.

As shown in the previous chapter, the decay of a given room's sound field is exponential, therefore, a precise method is to calculate T_{60} from the part of the impulse response that behaves exponentially. The difference between the measurement signal and noise power corrupting the measurement signal is the signal to noise ratio (SNR). It is defined as [18]

$$\text{SNR} = \frac{P_{signal}}{P_{noise}}, \tag{23}$$

where P is average power. In room-acoustic measurements P_{noise} is background noise caused by traffic, ventilation and other activities happening in the room that is being measured. In practical room acoustic measurents, SNR is rarely 60 decibels as is required by the Sabine equation, therefore the calculation of T_{60} is extrapolated from a 20 or 30 decibel decay.

The method of calculating acoustic energy decay from a room impulse response is called the backwards Schroeder integral, which is defined as [26]:

$$S(t) = \int_t^\infty p^2(t)\mathrm{d}t = \int_\infty^t p^2(t)d(-t), \tag{24}$$

where $p(t)$ is the impulse response. When the sound source is turned off, t is zero. The integration produces an even decay curve, to which a straight line can be fitted and the value of T_{60} can be calculated, as shown in Figure 3.

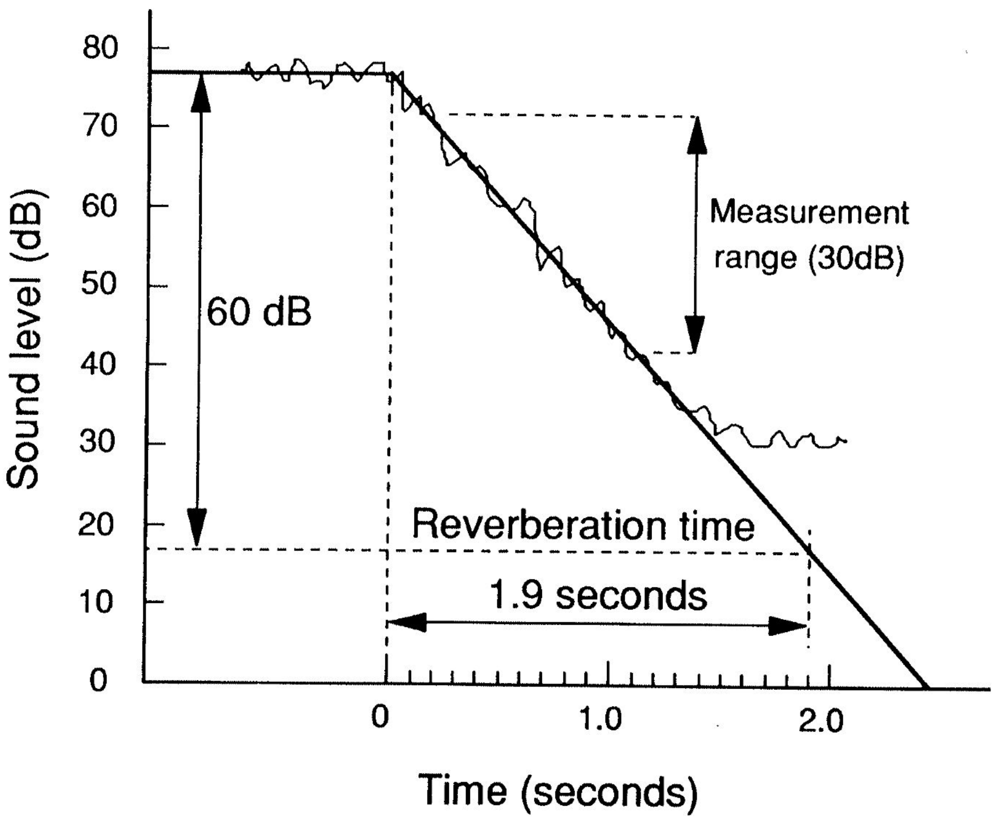

Figure 3: Reverberation time extrapolated from a 30 decibel decay (adopted from [5])

Room modes interfere with the measurement regardless of analysis method. Room modes decay at their unique rates and so cause irregularities in the decay curve. There are also other sources of errors, which will be covered in Chapter 3.2

3.1.1 Measurement Signals

In order for room impulse responses to be obtained, a sound source emitting a measurement signal is needed. That signal has to be recorded in order to calculate the reverberation time as described above. Common measurement signals include:

- Interrupted noise
- Sine-sweep signal
- MLS-signal (maximum length sequence)
- Impulse-like signals created using air-balloons, starter gun etc.

The interrupted noise and the impulse-like noise methods are quite close to Sabine's original method. The room is acoustically excited using a wide-band signal, which is recorded inside the room and then analyzed. These methods provide decay curves directly via the measured signal. However, the interrupted noise method does not provide an impulse response, only reverberation time. When performing reverberation time measurements using the interrupted noise method, it is very important to make sure that the build up time has passed before the sound source is turned off again.

With impulse-like sound sources such as gunshots, exploding balloons, handclaps or such, exact repeatability of the excitation is always an issue. These signals are usually only used in cases where none of the other methods are available.

Although the two aforementioned methods have well known limitations they are allowed to be used during standardized absorption testing according to ISO-354 [3].

The two other methods are more error tolerant methods that are generally more suitable for measuring impulse responses. In these methods the recorded signal is not directly a decay curve and so the impulse response must be found by processing the recorded signal.

Maximum Length Sequences

A maximum length sequence is a pseudorandom and deterministic signal which is fed to a system, recorded and analyzed by cross-correlation. This is to say that contrary to the other methods, MLS signals are processed in time domain. The cross-correlation process is circular as the signal sent to the system is known. This means that with special processing, a variant of the Fourier Transform called the Fast Hadamard Transformation, the impulse response can be calculated almost as fast as with regular Fast Fourier Transform (FFT) even when working in time-domain.

MLS-signals have a few advantages in comparison to the unprocessed measurements. Because of the determinant nature of the measurement signal, the measurement method allows for maximum repeatability. The MLS-signal is usually pseudorandom white noise, making its' signal-energy quite high, which means it has a good signal-to-noise ratio (SNR).

An MLS-measurement does not suffer from any windowing or cutoff-effects due to the fact that an MLS-signal is periodical and can be analyzed one period at a time. An MLS-measurement can be repeated in such a way that the measured signal data is summed to itself in phase. This theoretically increases the SNR by 3 dB every time the number of measurements is doubled, as the tradeoff is the measurement duration.

The MLS-method also introduces criteria for the measurement apparatus. The whole system in which the MLS-method is used has to be linear and time-invariant. Linearity introduces limitations for loudspeaker distortion. Using today's technology, the time-invariance requirement does not effect the sound system but the room itself. For example the temperature of the room should be constant throughout the measurement. Disturbances can also be caused by moving air due to ventilation or an open window, for instance.

Logarithmic Sine-Sweep

The logarithmic sine-sweep method is a relatively simple implementation that outperforms all the other methods described above [25]. The measurement uses a sinusoidal signal with a frequency which varies exponentially over time. The sine sweep, $x(t)$, is recorded in the room and the recorded sound is $y(t)$. $x(t)$ can be "packed" into a unit impulse function by convolution [10]:

$$x(t) \otimes f(t) \Rightarrow \delta, \tag{25}$$

where $f(t)$ is an inverse filter. To obtain the systems impulse response, the measured signal $y(t)$ is convoluted with the inverse filter $f(t)$ [10]:

$$h(t) = y(t) \otimes f(t), \tag{26}$$

where $h(t)$ is the systems impulse response in the time domain.

The frequency changes more slowly at the start of the signal, therefore containing more power at lower frequencies, which gives the sweep an unintended equalization. This has to be taken into consideration when calculating the impulse response.

The major advantage of the logarithmic sweep method is that due to the convolution process, all harmonic distortion components are separated at the end of the impulse response. A linear convolution "packs" the response using a filter with a group delay, that follows the measurement signal. In other words, if a loudspeaker has produced a second order harmonic distortion component for example, it will appear on the other side of the actual response from where it can be windowed to leave only the actual response [25].

This makes the logarithmic sweep method a suitable method for measuring spaces that have background noise or are not always time-invariant. Because of its computational simplicity, the logarithmic sweep method is commonly implemented in most audio processing software, even ones which are not commonly used in acoustical measurements.

3.1.2 Measurement Equipment

Microphones

The standard for absorption measurements requires that omnidirectional condenser microphones are used when measuring reverberation time. Omnidirectional microphones react to sound waves from all incident angles within a limit of sensitivity required by the used method. Most room acoustic measurements are performed using omnidirectional microphones. Figure 4 shows the polar response pattern of a DPA 4053 omnidirectional microphone [7]. There are a few special cases where microphones with more narrow patterns are used, but they will not be discussed in this thesis.

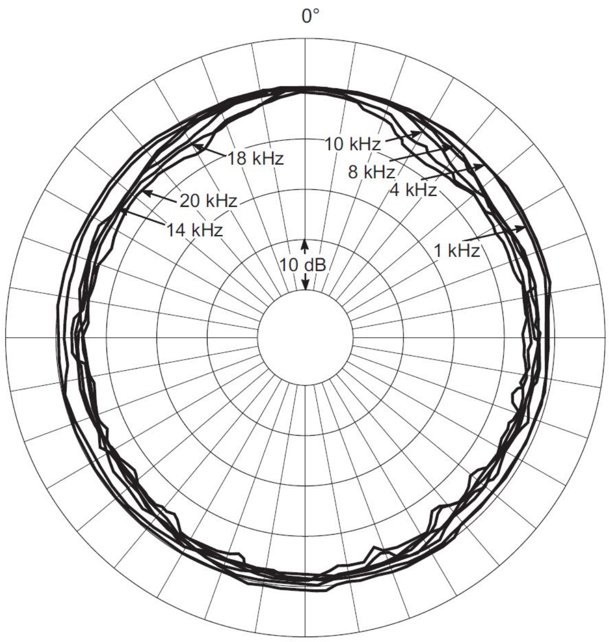

Figure 4: Polar response of a DPA 4053 microphone (adopted from [7])

Loudspeakers

For room acoustics measurements an omnidirectional loudspeaker is most commonly used, these loudspeakers are usually constructed with the elements placed on each side of a polyhedron structure, usually a dodecahedron. Figure 5 shows a loudspeaker with elements on each side of a dodecahedron.

Figure 5: Dodecahedron loudspeaker (adopted from [19])

These loudspeakers are omnidirectional at lower frequencies but at higher frequencies the individual elements can be located from the loudspeaker's polar pattern. The standard for reverberation measurements specifies the use of omnidirectional loudspeakers with a maximum acceptable deviation in sound pressure over a gliding average of 30^0 arcs in free field. Table 1 shows the maximum amount of deviation allowed [13]:

Table 1: Maximum deviations for an omnidirectional speakers' frequency response

Frequency, Hz	125	250	500	1000	2000	4000
Maximum deviation, dB	± 1	± 1	± 1	± 3	± 5	± 6

3.1.3 Absorption Coefficient Calculations

Acoustic absorption can be calculated using the difference in measured reverberation times between the T_{60} of an empty room and T_{60} of the same room with the absorptive material in place. The absorption of the test specimen is then [3]

$$A_{\mathrm{T}} = 55,3V\left(\frac{1}{c_2 T_2} - \frac{1}{c_1 T_1}\right) - 4V(m_2 - m_1), \tag{27}$$

where A_{T} is the equivalent sound absorption area of the test specimen. The Sound absorption coefficient is calculated as follows [3]:

$$\alpha_s = \frac{A_{\mathrm{T}}}{S}, \tag{28}$$

where S is the area covered by the test specimen.

In reverberation chamber measurements, omnidirectional sound sources are also used. A reverberation chamber is a static laboratory environment.

3.2 Standardized Measurement Methods

The primary goal of material acoustic measurements is to find the ratio between absorptive and reflective material in a room, in other words to measure acoustic absorption, which describes a materials' behaviour in a given space. There are three standardized methods of measuring acoustic absorption:

- The reverberation-chamber method, ISO-354:2003
- Impedance-tube measurement using the standing wave ratio method ISO-10534-1:2001
- Impedance-tube measurement using the transfer function method ISO-10534-2:2001

3.2.1 ISO-354 Reverberation Chamber Measurement

Measurement Principle

The reverberation chamber method is performed as a comparison measurement between the reverberation times of an empty chamber and with the absorptive material in place. The absorption coefficient is calculated using the formulae presented in Section 3.1.3

Observations

The reverberation chamber method is a standardized method, in which limits and directions are clearly defined in order to minimize uncertainties and errors.

The size of the test specimen depends on its physical properties and chamber volume. For reverberation chambers with $150 \geq V \leq 200$ m^3, if the material seems to be absorptive, the test specimen has to cover an area of 10 m^2 and in cases where the maetrial seems to be reflective, an area of 12 m^2 is required. These sizes however, are used as guidelines due to the fact that some materials may only be available as modules of predetermined size. If the chambers' volume is over 200 m^3 then the required area's upper limit is encreased by a factor of $(V/200 \text{ m}^3)^{2/3}$. These rules apply when the test specimen can be installed on a plane.

The reverberation chamber method also applies for discrete objects such as chairs or desks. These objects are installed in the same manner as their purpose of use.

One important consideration with this method is that total diffusivity of the sound field within the chamber is assumed. On a practical level, this is not possible to achieve, but the standard allows the measurement if the condition of diffusivity is met. The test defines that sufficient diffusivity is achieved when the acoustic absorption of the space no longer increases by the addition of acoustic diffusers into the chamber.

The atmospheric conditions of the chamber should be constant throughout the entire measurement process. As shown in Section 2.1.1, humidity and temperature affect air traversal. The measurement conditions required are as follows:

- Relative humidity in the chamber should be at least 30 % and at most 90 %
- The temperature in the room should be at least 15^oC

Measurement conditions may vary significally especially if measurements are performed in conditions where indoor and outdoor temperatures differ greatly from one another, for example indoor measurements carried out during winter. To avoid measurement errors, all specimens should be allowed to reach equilibrium with the room conditions before measuring. An automatic temperature/humidity logger is not a necessity, but is helpful.

3.2.2 ISO-10534-1 Impedance Tube Measurement - Standing Wave Ratio

Impedance-tube measurements describe methods where the sound field inside a tube is measured. A sample material is attached to one end of a straight, rigid and smooth tube and a loudspeaker is placed at the other end of the tube. Microphones are attached so that they can be moved lengthwise inside the tube. The loudspeaker produces a static sound field inside the tube, which is then measured and analyzed.

Both impedance-tube methods produce absorption coefficients for 0 angle sound incidence only.

Theory

In the aforementioned situation, the loudspeaker produces a standing wave inside the tube. The impedance of the materials surface is defined as the ratio between sound pressure and particle velocity

$$\mathbf{Z}_n(f) = \frac{\mathbf{P}(f)}{\mathbf{U}(f)}, \tag{29}$$

Microphones measure sound pressure precisely, however, at present, a measurement device for measuring particle velocity has been invented, but has not yet been introduced into use for this method. Particle velocity can therefore be calculated from sound pressure measurements within the tube [8]

$$u_i(x) = \frac{1}{Z_0} p_i(x), \tag{30}$$

$$u_r(x) = -\frac{1}{Z_0} p_r(x), \tag{31}$$

where Z_0 is the impedance of a plane wave. A reflected plane wave can be written as a function of the reflection factor as follows [8]:

$$p_r(x) = r p_0 e^{-jk_0 x}, \tag{32}$$

where $p_i = p_0 e^{-jk_0 x}$, a complex presentation of the incident wave. The incident wave is assumed to be planar, harmonic at f and it is also assumed not to dampen inside the tube. When $x = 0$ [8],

$$\mathbf{Z} = \mathbf{Z}_0 \frac{1+r}{1-r}, \tag{33}$$

and [8]

$$\mathbf{r} = \mathbf{Z}_0 \frac{(\mathbf{Z}/\mathbf{Z}_0) - 1}{(\mathbf{Z}/\mathbf{Z}_0) + 1}, \tag{34}$$

Acoustic absorption can be calculated from [8]

$$\alpha = 1 - |\mathbf{r}|^2. \quad (35)$$

Measuring Principle

Impedance is a function of frequency, in this case, a frequency response as a ratio between sound pressure and particle velocity as stated in Equation 29. The method measures the response one frequency at a time. Figure 6 shows the acoustic phenomena in the tube where the loudspeaker produces a plane wave, p_i, which together with the reflected wave, p_r, produce a standing wave, $p = p_i + p_r$.

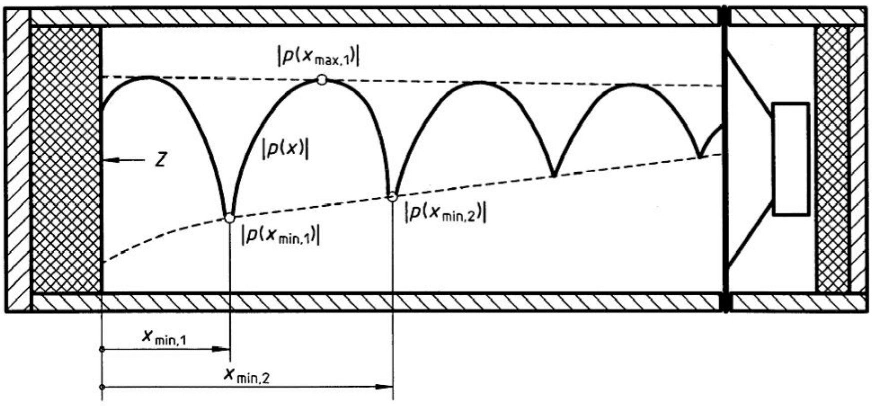

Figure 6: Standing wave method (adopted from [8])

The principle is to measure the pressure maxima and minima, $|p(x_{\text{max}})|$ and $|p(x_{\text{min}})|$ respectively and their distances from the sample material. With this information, acoustic absorption can be calculated. Additionally, the distance from the reference point $x = 0$, usually the materials surface, to the first pressure minimum, $x_{\text{min, 1}}$, and its wavelength, λ_0, is also measured in order to find out the materials reflection factor, r.

A pressure maximum occurs when p_i and p_r are in phase and a pressure minimum occurs when p_i and p_r are out of phase. The ratio of a standing wave is [8]:

$$s = \frac{|p(\text{max})|}{|p(\text{min})|}, \quad (36)$$

which can be written [8]

$$s = \frac{1 + |r|}{1 - |r|}, \quad (37)$$

and [8]

$$|r| = \frac{s-1}{s+1}, \tag{38}$$

and the acoustic absorption of the material can be calculated using Equation 35.

3.2.3 ISO-10534-2 Impedance Tube Measurement - Transfer Function Method

The main difference between the transfer function method and the standing wave ratio is the measurement of the acoustic phenomena occuring inside the tube. The standing wave ratio method uses one frequency at a time whereas the transfer function method can measure the whole frequency band at once. This means that the measurement signal can be wide range, such as that described in Section 3.1.1.

Measuring Principle

Transfer function method measurements use the same prerequisites as the standing wave ratio measurements. The sample material is attached to one end of a tube and a loudspeaker is mounted to the other end and inside the tube is a microphone to measure the occurring acoustic phenomena.

In this method, the complex transfer function, $\mathbf{H}_{12}$, between two microphone points are measured in order to calculate the same acoustic descriptors as in the previous method.

This measurement can be performed using one or two microphones. Both methods have practical challenges. Using one microphone requires moving the microphone between two measurement points but it eliminates an unavoidable measurement error that originates from the differences in a microphone pair. Using one microphone also allows a measurement point to be optimally chosen for a specific frequency range.

A stationary signal is used as a measurement signal. The signal can be random, pseudorandom or the logarithmic sweep. When one microphone is used, the time that is required for changing it's position must be dissipated, meaning that the measurement signal must be deterministic. Pseudorandom noise works best when perfoming the measurements using this method.

When using two microphones, they can be attached to the edges of the tube, making the test sample the only moving part of the measurement. The standard that defines this measurement [9] recommends the use of two microphones. This method is illustrated in Figure 7.

In Figure 7 objects 1 and 2 show the microphones and object 3 is the test sample. The microphones used in this measurement must be a matched pair. When microphones are mounted on the side wall of the tube, regular omnidirectional microphones can be used.

The transfer function between the microphones is [9]

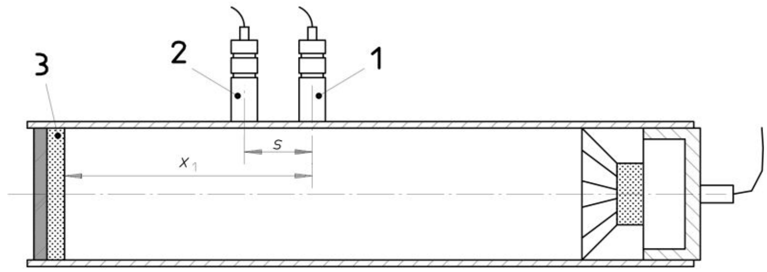

Figure 7: The transfer function method using two microphones (adopted from [9])

$$\mathbf{H}_{12} = \frac{\mathbf{S}_{12}}{\mathbf{S}_{11}}, \tag{39}$$

where $\mathbf{S}_{12}$ is the cross spectrum between the two microphones $\mathbf{S}_{12} = \mathbf{p}_2 \cdot \mathbf{p}_1^*$, where $\mathbf{S}_{11}$ is the autospectrum of the first microphone $\mathbf{S}_{11} = \mathbf{p}_1 \cdot \mathbf{p}_1^*$. The reflection factor can be calculated from the measured transfer function [9] :

$$r = \frac{\mathbf{H}_{12} - \mathbf{H}_I}{\mathbf{H}_R - \mathbf{H}_{12}} e^{2jk_0x_1}, \tag{40}$$

where $\mathbf{H}_I$ is the transfer function for the incident wave alone and $\mathbf{H}_R$ is the transfer function for the signal alone. $\mathbf{H}_I$ and $\mathbf{H}_R$ is measured between the two microphone positions. x_1 is the distance between the test sample and the first microphone and k_0 is the wave number, which is defined as

$$k_0 = \frac{2\pi f}{c_0}. \tag{41}$$

Acoustic absorption can be calculated using Equation 35

3.2.4 Measurement Frequencies

The goal of all absorption measurements, regardless of method or technique, is to reliably measure acoustic absorption on a frequency-range within third octave bands with centre-frequencies shown in Table 2 [1]

Table 2: Third octave band centre-frequencies

100	125	160	200	250	315
400	500	630	800	1 000	1 250
1 600	2 000	2 500	3 150	4 000	and 5 000 Hz

A third octave band is part of a frequency range, which is defined geometrically by dividing an octave band into three equal parts. An octave band is defined by f_1, being the low limit, and the high limit, which is twice that of f_1. A third octave band is defined as [18]:

$$f_2 = \sqrt[3]{2} f_1. \tag{42}$$

Measurements can be performed outside of the range shown in Table 2 but, especially with reverberation chamber measurements, below 100 Hz room modes tend to dominate the sound field making the results unreliable. The Schroeder-frequency of a normal reverberation chamber is usually close to 100 Hz. Measurements can also excede 5 kHz, but the attenuation of sound due to its' propagation in the medium is relatively large and a small error in the use of correction terms may cause further inaccuracies in the results.

In the standing wave measurements the frequency range is defined by the length of the testing area, in other words, the distance between the loudspeaker and the edge of the installed test sample. The length of the test area is defined as [8]:

$$l \geq 3\lambda_0/4, \tag{43}$$

where l is the length of the test area and λ_0 is the longest wavelength allowed inside the tube, therefore $\frac{c_0}{\lambda_0}$ marks the lowest measurable frequency.

Acoustically reactive materials, materials with very high reflectivity, tend to produce higher order harmonics in reflected waves. These harmonics are dampened within a length, three times that of the diameter of the tube. Thus, an impedance-tube should be twice as long as the aforementioned distance in order to avoid higher order harmonics from both the test sample and the loudspeaker transducer at the measurement point. The length of an impedance tube is defined as [8] :

$$l \geq 250/f_l + 3d, \tag{44}$$

where f_1 is the lowest measurement frequency, and d is the diameter of the tube and l is the tube's length. The upper limit of the frequency range is defined so that there is no change of transversal higher order modes inside the tube. These measurements require a strict one-dimensional sound field. For rectangular tubes [8]:

$$f_u d \geq 0.5 c_0, \tag{45}$$

and cylindrical tubes [8] :

$$f_u d \geq 0.58 c_0. \tag{46}$$

If the two-microphone method is used in the transfer function method, the distance between the two microphones proposes a new requirement for the upper limit of the working frequency range [9] .

$$f_u s < 0.45 c_0, \tag{47}$$

where s is the distance between the microphones. Using previously described formulae, the working upper frequency limit for tube measurements can be calculated. The standard recommends that s be at least 5% of the longest wavelength of the lowest measured frequency, assuming that the condition in Equation 46 is fulfilled. The low-frequency limit can therefore be calculated using the following equation:

$$f_{l(mic)} = \frac{0.05 * c_0}{s_{mic}}, \tag{48}$$

where s_{mic} is the distance between the two microphones.

4 Alternative Methods

4.1 Why Alternative Methods Exist?

Alternative measurement methods are developed in order to investigate whether the acoustic absorption can be reliably measured without the restraints of the standards, as they have been developed mainly out of practical necessity. In situations where it is necessary, for example, to measure the acoustic absorption of seats in a protected buildings' concert hall, they can not be removed and measured. The acoustic absorption of the seats must therefore be measured inside the hall itself, which creates a problem: For situations like this, no standardized method for acoustic measurements exist.

Methods have been developed for situations when using a standardized method is not an option. However, these methods always compromise measurement accuracy, frequency range and repeatability but as they may sometimes be the only available option, the search for such methods is of great interest especially to acousticians involved in concert hall renovations. These alternative methods may include measurements performed "in situ" or the use of scale models.

4.2 Alternative Methods Reviewed For This Thesis

Two alternative measurement methods will be presented in this thesis; a scale model measurement and a reverberation measurement inside a room that does not exactly meet the standard of a reverberation chamber, but is a reverberant space with low background noise.

4.2.1 Scale Model Measurement

The standard for reverberation chamber measurement [3] defines the limits for chamber volume, but the same measurement principle is applicable for accurate measurements in scale models. Scale models are commonly used for designing concert halls. A scale model could for example, be a tight plexiglass construction with an irregular shape.

Measuring in small scale is a special case of the standardized reverberation chamber measurement. Models rely on the laws of acoustical similarity; in which all sound phenomena are comparable to full scale situations when the frequency range is scaled up according to the model scale [23].

Small scale models were the most common design tool for concert halls before computer modeling was generalized. The main advantage of computer modeling is that changes in the design can be implemented quickly in comparison to scale model building. The largest disadvantage of computer modeling is its inaccuracy for calculations of sound scattering, reflection and diffraction. These values are always approximated in a computer simulation, thus resulting in errors in various concert hall descriptors. In larger concert hall projects both scale modeling and computer simulation are used as design tools. Studies [24] have shown that results from scale models and computer simulations are cohesive.

Measuring low frequencies inside a model is not possible because the sound field is not diffuse. As in full scale, the modes of a room affect its frequency response. A Schroeder-frequency can be calculated for a small scale room. The lowest possible frequency inside a small scale room is half the wavelength that fits between the longest distance between two walls. Below that frequency the room functions as a pressure chamber. In any case, the measurement is reliable only over the Schroeder-frequency.

The small scale measurement could provide full scale results if the measurement conditions are transposed to match the small scale situation. For example, the model could be scaled to 1:10 the size of a reverberation chamber. This means that the measured frequency range should be scaled up to match the small scale situation; 100 Hz scales up to 1 kHz, 1 kHz scales to 10 kHz etc.

Loudspeaker selection is crucial when performing small scale measurements. The loudspeaker has to have a flat response in the measured range. Regular loudspeakers usually used for acoustical measurements have a flat response up to 10 kHz. High intensity impulse-like noise at high frequencies can be produced using spark discharge-impulses. [29]

The attenuation of air is not linear. For frequencies above human hearing ($\geq$ 20 kHz) attenuation is so extreme that sound simply will not propagate long enough for natural decay to occur. To compensate for air absorption, one or more of the following methods are used:

- The medium inside the model is changed. It is common practice to switch air to nitrogen when performing scale model measurements
- The effect of air attenuation can be negated numerically using correction terms according to [4]
- The measurement signal can be amplified frequency-dependently to account for attenuation of air

Studies [22] show that it is impossible to accurately account for air absorption. This problem has been thoroughly analyzed in previous research.

Measurement of reverberation time in a scale model is possible to perform. However, measuring absorption correctly is a challenge. When measuring scale models of actual concert halls, the materials used in the scale model are not representative of the materials used in the actual hall. If the material is absorptive, then measurements of acoustic absorption can be performed [23]. This is also the case when scale modeled audiences are measured. Figure 8 shows a model audience inside a small scale reverberation chamber.

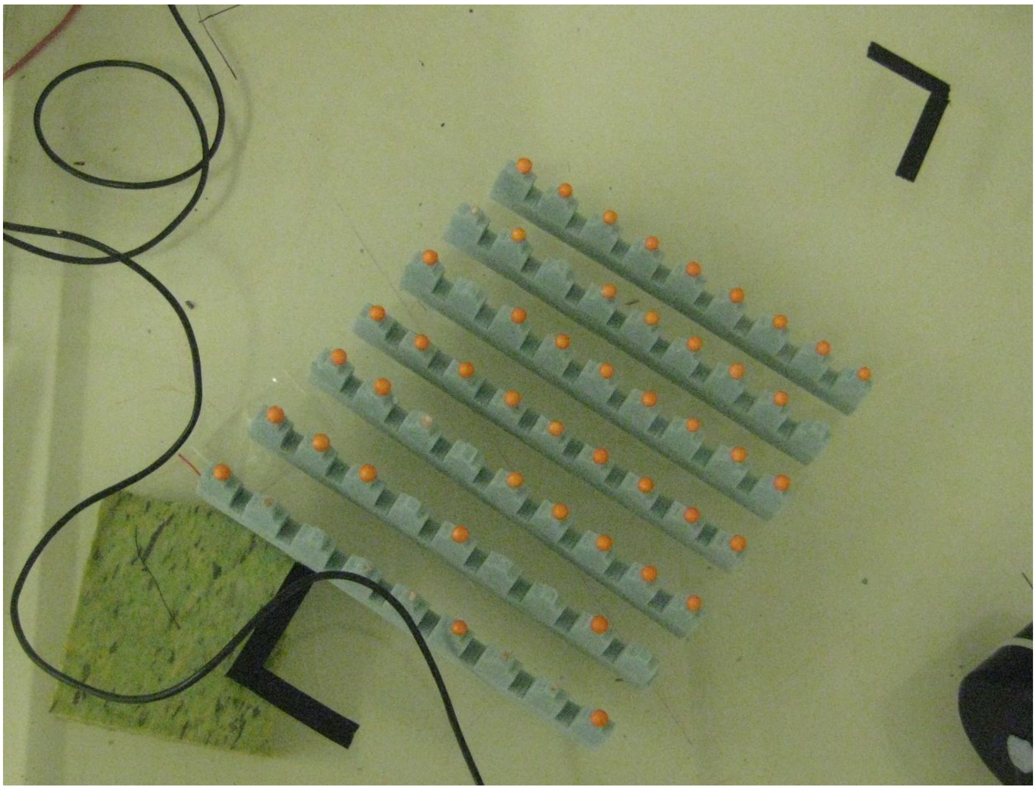

Figure 8: Scale modeled audience

4.2.2 Measurement in a Reverberant Space

Laboratory measurements for acoustic absorption should fulfill two basic requirements:

- It should describe the basic concept of absorption, by accurately representing the behaviour of the the sound field when the material is installed inside the room
- A certain level of accuracy. The repeatability factor according ISO-354 to should be 0.05 or less. [3]

In a reverberant space that does not qualify as a reverberation chamber, it is still possible to achieve conditions that favour measurements. The level of background noise must be low enough to enable a decent signal to noise-ratio to be achieved. When the room volume is smaller than the requirement of the standard, its Schroeder-frequency will determine the lowest possible third-octave band for which the measurement will be accurate.

Diffusivity of a rectangular shaped, empty room should be discussed as well. The diffusivity-test described in [3] states that volume diffusers should be added until the absorption coefficient no longer increases. Diffusers in a smaller empty space might already create too much absorptive surface within the room, therefore, diffusivity of the room may not be fully achieved in spaces of all sizes. Because of its nature, this measurement method must compromise diffusivity in order for comparable reverberation times, between the empty room and the room with the material in place to be measured. If the reverberation time decreases too drastically in the empty room, the difference in the reverberation times will be too small for analysis.

The diffusivity problem described above can be solved by adding source and receiver points within the room. The ISO 140-13 standard for building acoustics states that rooms with a volume under 250 m^3 should be measured using 18 independent measurement points [12]. The only compromise is the measurement duration.

5 Methodology and Results

5.1 Measurement Pre-Requirements

For room measurements to be performed correctly, the omnidirectionality of two loudspeakers was tested. As briefly described in Chapter 3, if the loudspeakers meet the requirements of ISO-3382 , they can be used in reverberation room measurements [13]. This section will also describe the material that was measured in all the measurement methods previously described.

5.1.1 Loudspeakers

For room acoustic measurements, the omnidirectionality of two loudspeakers were measured at the acoustics laboratory at the Aalto University. The measurements were performed on the second of February 2011. The measured loudspeakers were two cubes of different sizes with an element arranged on each surface (shown in Figures 9 and 10). The frequency response of the loudspeakers was measured in anechoic conditions. The large loudspeakers' edge measured 282 mm and the small 173 mm. The 6-inch elements mounted on the larger loudspeaker were 18Sound 6ND 430. The smaller loudspeakers' 4-inch elements were VIFA MG10SD09.

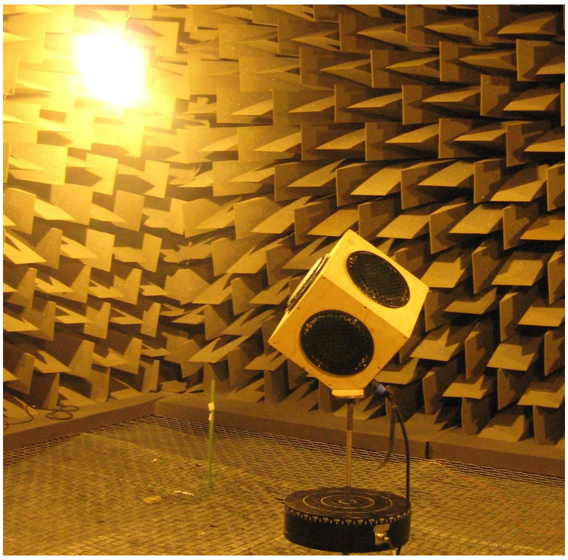

Figure 9: The large loudspeaker on the turntable inside the anechoic room

A microphone was positioned to point at each loudspeaker at a distance of 2.4 meters. The loudspeakers were fed a logarithmic sine sweep signal at a length of 128 000 samples. The measurement equipment consisted of a laptop computer which ran the measurement software. A Marian Ucon CX - USB sound card was used as an A/D-converter. The sound cards internal sample rate during the whole measurement process was 48 kHz. The measurement signal was both generated

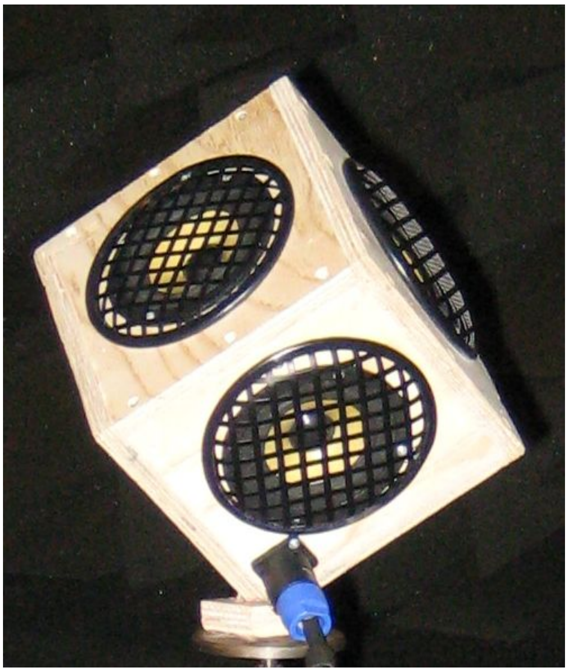

Figure 10: The small loudspeaker inside the anechoic room

and recorded using the ARTA software. A Crest Audio CA 6 amplifier was used and a B&K 4006 omnidirectional microphone was used. Most of the measurement equipment can be seen in Figure 11; the sound card at the far left, the laptop and the amplifier in the center and the turntable's control unit on the right.

Figure 11: The measurement equipment

To ensure the "worst possible case" [2], both loudspeakers were mounted by their corners on a turntable at the center of the anechoic room. The directional pattern was measured by measuring impulse responses at five degree intervals. The omnidirectionality of the loudspeakers can be calculated from the same set of results.

The loudspeakers were measured in anechoic conditions at five degree intervals, from which polar patterns were calculated. From the same set of measurements, omnidirectionality requirements of ISO-3382 and ISO-140-4 were calculated [13] [2].

Polar Patterns

The polar responses were calculated and plotted and shown in Figures 12, 13,14 and 15. Figures 12 and 14 show, that on low frequencies both loudspeakers have a flat polar pattern, but only the larger loudspeaker has a relatively flat frequency response. Figures 13 and 15 show, that at high frequencies, individual elements of the loudspeaker start to distinguish themselves in the polar pattern, but both loudspeakers have an uneven frequency response.

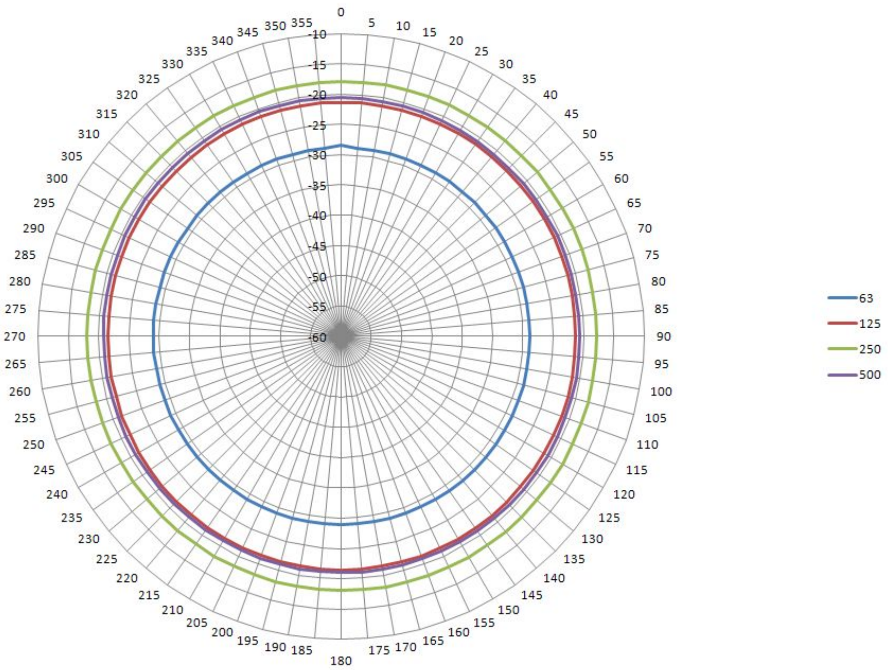

Figure 12: The polar pattern of the large loudspeaker (low frequencies)

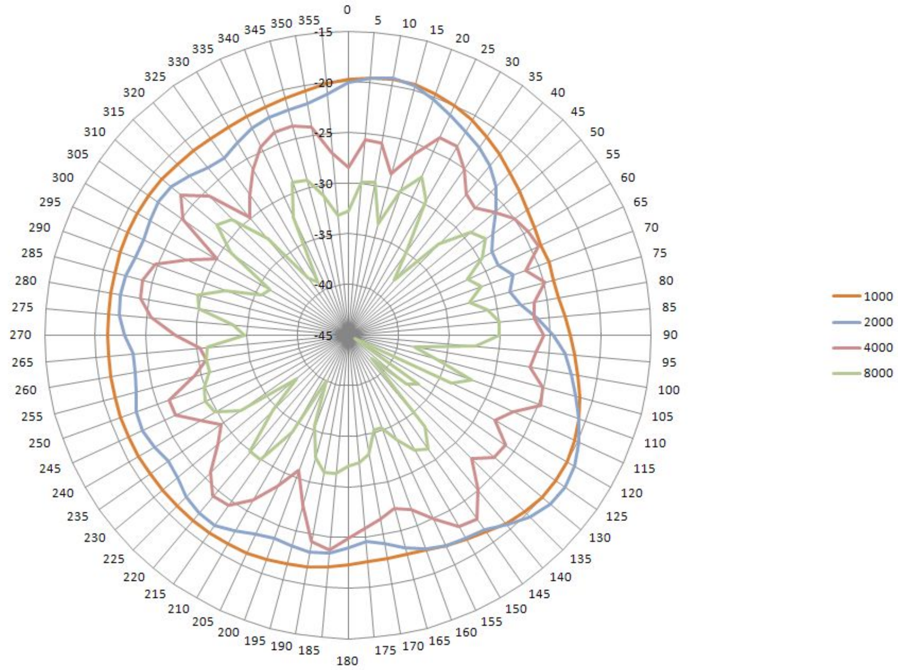

Figure 13: The polar pattern of the large loudspeaker (high frequencies)

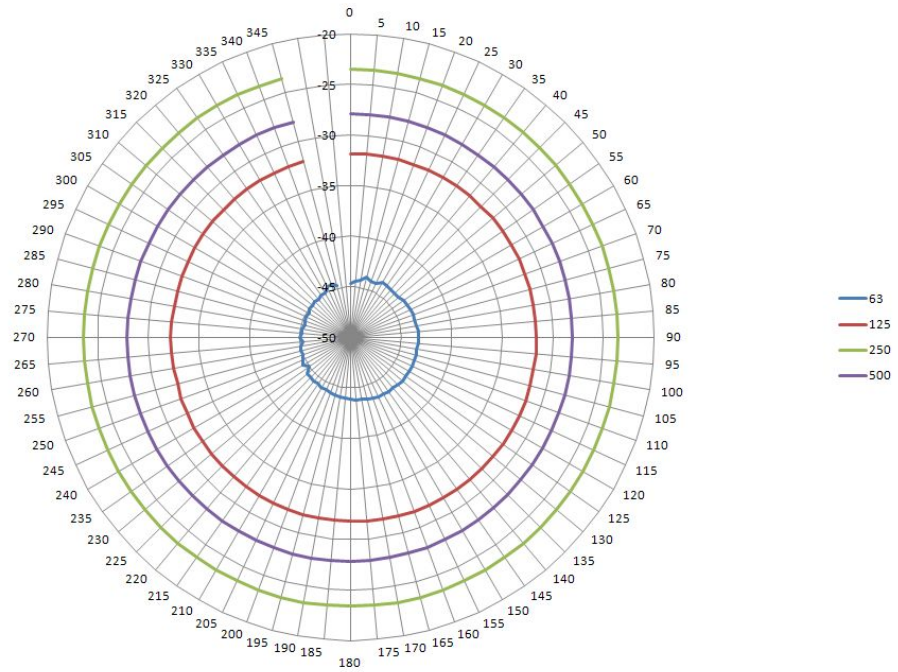

Figure 14: The polar pattern of the small loudspeaker (low frequencies)

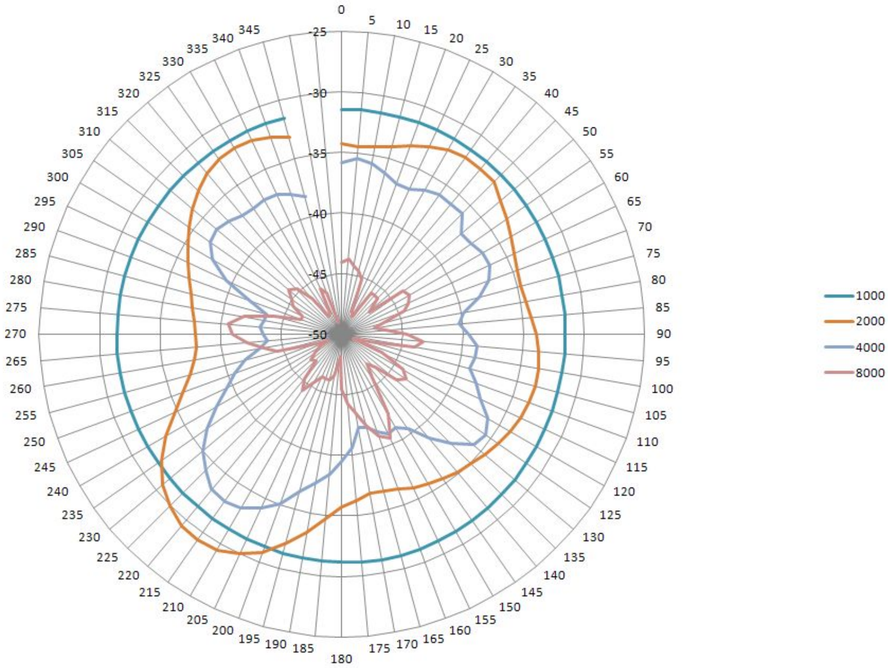

Figure 15: The polar pattern of the small loudspeaker (high frequencies)

5.1.2 Material

A glasswool board, Ecophon Master C, of size 0.6 m $\times$ 1.2 m $\times$ 0.04 m was measured using the tube, room and small scale measurements. The board with its corner cut off is shown in Figure 16. Ecophon provided data from measurements carried out on the Master C board, using the reverberation chamber method. These results will serve as a basis for data analysis throughout the rest of this thesis.

The acoustic absorption, according to ISO-354 in third octave bands for the Master C board are presented in Table 3 [3].

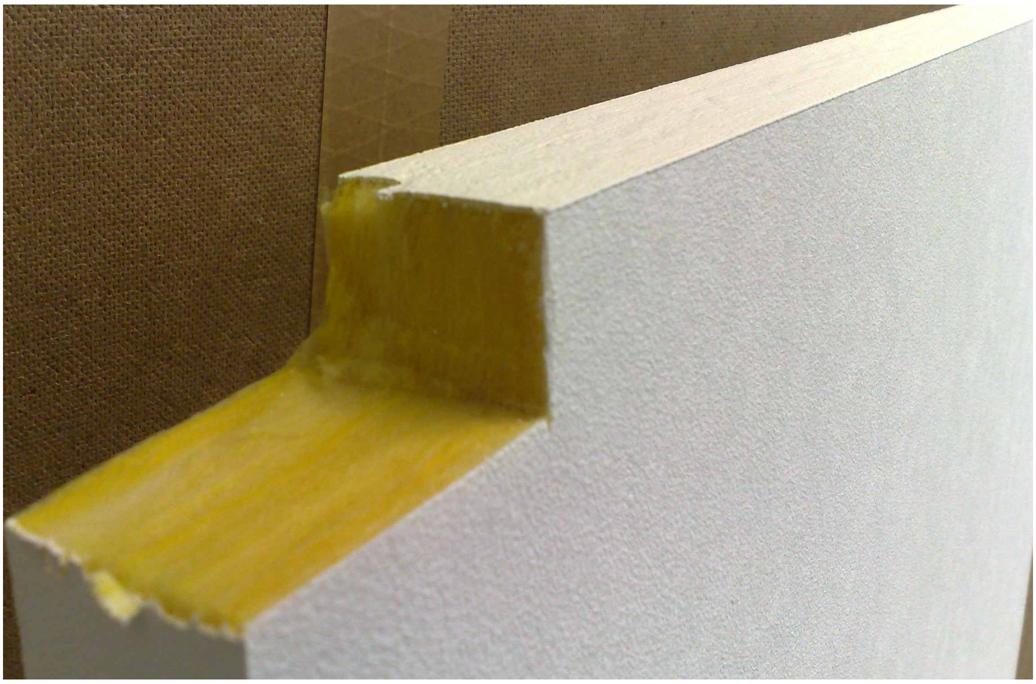

Figure 16: The Ecophon Master C glasswool board

Table 3: Measurement results of Ecophon Master C according to ISO-354

Thickness	40mm
Surface Area	10.8 m^2
Room Volume	200 m^2
Absorption	
Frequency (Hz)	α_s
50	0.01
63	-0.01
80	0.04
100	0.10
125	0.17
160	0.38
200	0.53
250	0.76
315	0.89
400	1.02
500	1.02
630	1.02
800	1.00
1000	0.98
1250	0.98
1600	0.96
2000	0.95
2500	0.98
3150	0.96
4000	0.95
5000	0.91

5.2 Measuring Acoustic Absorption

In this section, the procedures carried out for measuring acoustic absorption, using the methods described earlier, will be described. This section will cover the reverberation room, scale model and impedance tube methods.

5.2.1 Reverberation Room

An empty room of size 5.9 m $\times$ 4.8 m $\times$ 2.6 m was used as a reverberant space. The volume of the room was 73.6 m^3. The room had hard surfaces, which made it suitable to test this method. The reverberation room is shown in Figure 17.

The room modes of the reverberation room are shown in Figure 18. The modes were calculated using Equation 21, which is an approximation, because the room

Figure 17: The reverberation room with the sample material installed

surface-materials are never uniformly distributed. Therefore, Figure 18 shows the approximate frequencies where the modal density is high and low, which will be more interesting in the analysis phase, how the modes may effect the results.

An estimation of the room's Schroeder frequency was calculated using Equations 22 and 18. The aforementioned equations can be rewritten as in order to estimate the Schroeder frequency of the room.

$$f_S = 2000 * \sqrt{\frac{0.161\frac{V}{A+4mV}}{V}}, \tag{49}$$

Sabines equation requires the absorption coefficient of the different surface elements, which were based on a material database for the Odeon room acoustics modeler. The room had a linoleum floor and a concrete ceiling. Of the walls three were made of concrete and one was made of chipboard. The Schroeder-frequency of the room was approximately 300 Hz.

In these types of measurements, the volume of the room determines the test sample area. The standard for the reverberation chamber method states that if a reverberation chamber has a volume greater than 200 m^3, the test sample area should be increased from 10 m ^{2} by factor $(V/200\text{m}^3)^{2/3}$. Using this factor, the coefficient for the reverberation room can be calculated to be 0.5. This would suggest that an area between 5 m^2 and 6 m^2 of the test sample should be installed into the reverberation room. The floor area of the room was 28 m^2 and the area of one board of test material was 0.72 m^2, therefore eight boards were placed on the floor to form a square in the center of the floor. The area of the square was 5.76 m^2. The reverberation

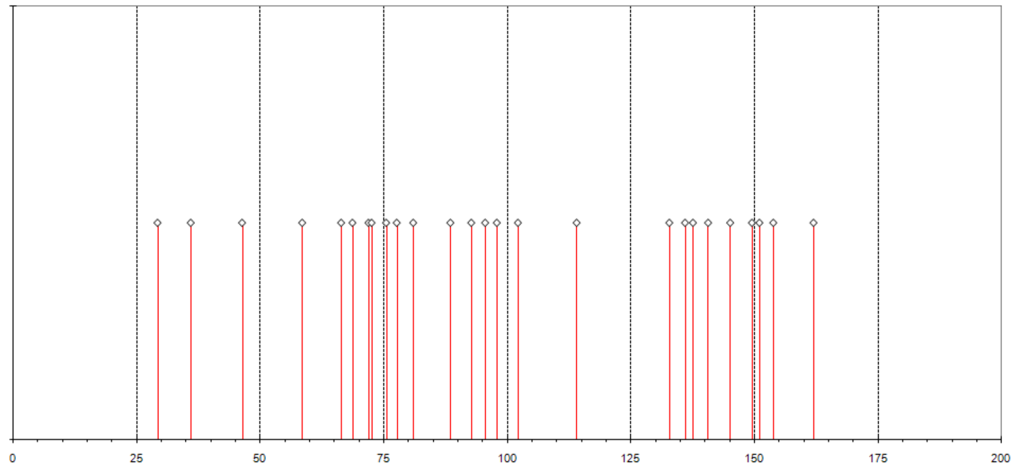

Figure 18: Modes of the reverberation room

room with the sample installed is shown in Figure 17.

The reverberation room measurement was performed according to ISO-354 [3]. To account for smaller room volume it was concluded that the measured reverberation times and therefore the acoustic absorption at under approximately 300 Hz were not reliable due to room modes. ISO-354 states that a sufficient approximation of the T_{60} inside a reverberation chamber can be measured using at least 12 independent measurement points [3]. ISO-140-13 on the other hand, states that an accurate T_{60} measurement for a room under the volume of 250 m^3 requires 18 independent measurement points [12]. A reverberation chambers volume can be between 150 m^3 and 500 m^3, and according to ISO-354, still be accurately measured using 12 measurement points.

The room was initially measured when empty using three source and six receiver points. Eight boards of the material were then installed and the measurements repeated. The material was installed on the floor, which meant that some edge effect was to be expected.

After the acoustic absorption was calculated, it was noted that the absorption curve had values over one in the reliable frequency range. The measurements were repeated using only six boards while trying to keep them as tight as possible to remove gaps between the boards. The measured results still showed absorption values over one. However, this was probably due to edge effect.

When the empty room was measured, room temperature was 20 ^{o}C and relative humidity was 20%. With the boards installed, the temperature remained at 20 ^{o}C but the relative humidity increased to 25%.

T_{60} Values

Figure 19 shows the plots of the minimums and maximums of measured reverberation times. As expected, the variation of results below the Schroeder frequency were quite substantial, but above that frequency, variations were much smaller.

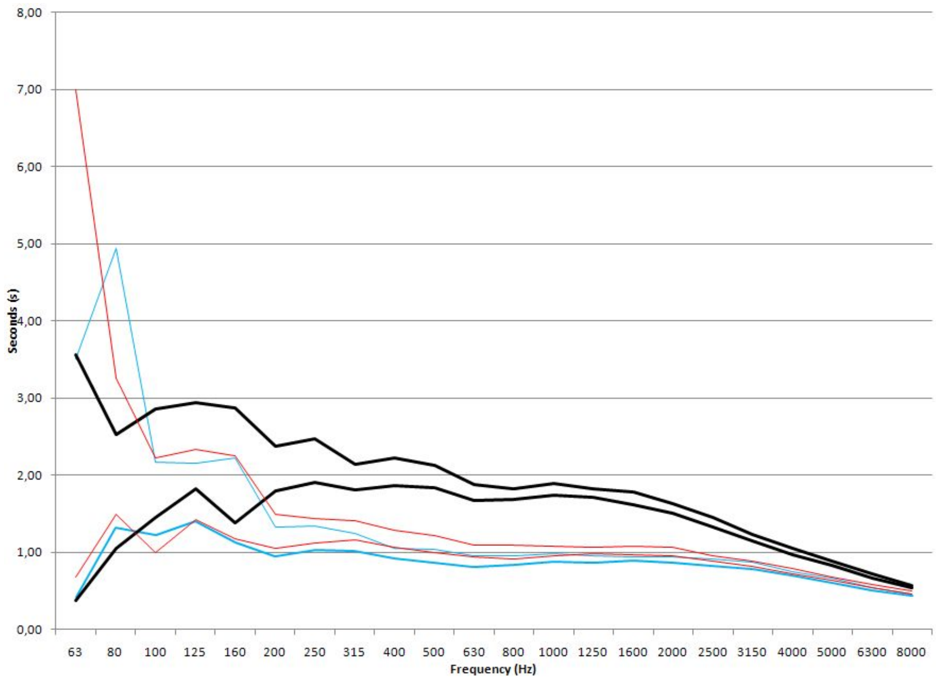

Figure 19: Minimums and maximums of measured reverberation times in the reverberation room

Acoustic Absorption

The acoustic absorption was calculated according to ISO-354 [3]. The reference values of absorption are plotted here also. As can be seen, at low frequencies room modes cause some variations in the values, but at higher frequencies the results are fairly accurate. Even at low frequencies the trend of the curve follows the reference curve quite well.

5.2.2 Scale Model Measurement

The small scale measurement was performed using an emptied concert hall scale model constructed using plexiglass. The volume of the scale model was 0.412 m^3. The scale model is shown in Figure 21. The measurement equipment and the framed material sample can be seen in Figure 21. The Schroeder-frequency of the model was approximately 2 kHz, which was calculated using Equation 49.

Absorption measurements inside the scale model were performed according to ISO-354 [3]. First the reverberation time inside the empty model is measured, then a

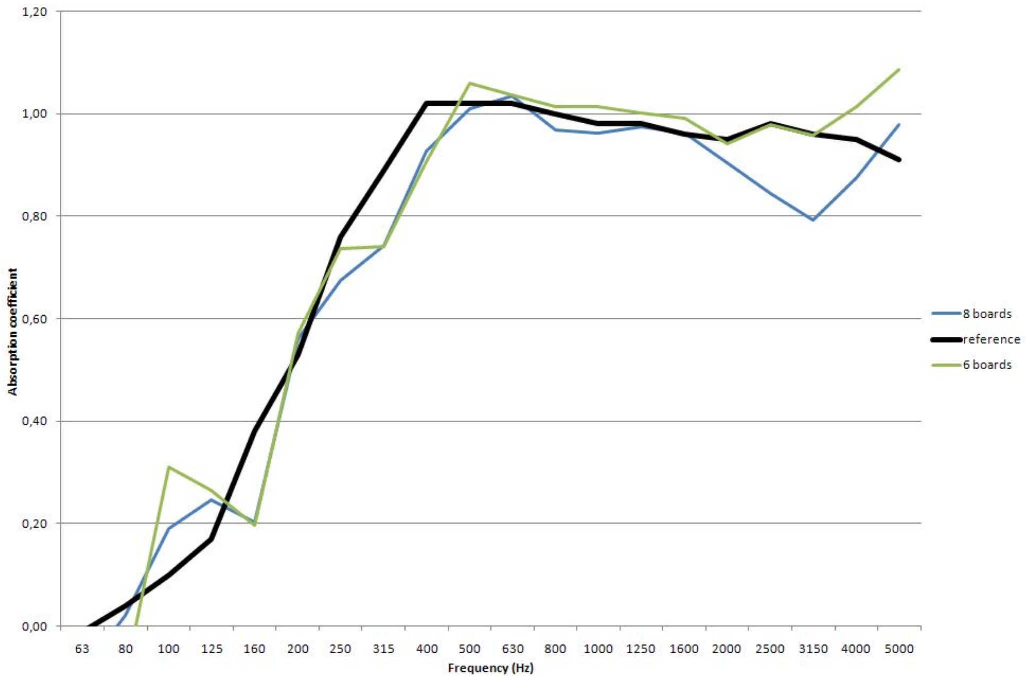

Figure 20: Acoustic absorption of the material measured in the reverberation room

material sample is installed and the reverberation time inside the model is measured. The acoustic absorption is then calculated according to ISO-354 [3].

The test sample used in the scale model measurement was framed using an acoustically reflective material. The purpose of the framing was to remove the edge effect.

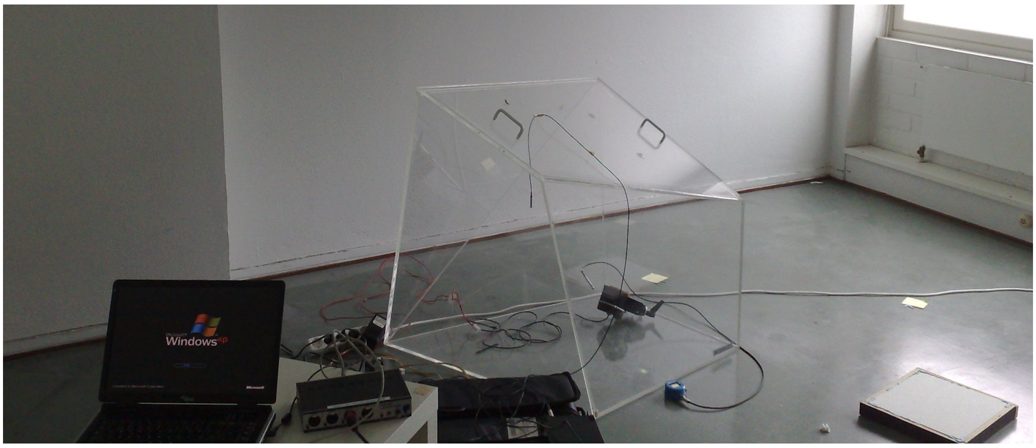

Figure 21: The scale model used for measuring acoustic absorption

The measurements were performed using 2 source positions and 6 receiver positions. The microphone was hung from holes in the sloping roof, making all receiver points independent. Three sets of measurements were performed inside the scale model, one measurement when the chamber was empty and two measurements when a material sample was installed. A logarithmic sweep was used as a measurement signal.

T_{60} Values

The measured reveberation times can be seen in Figure 22. Below the Schroeder frequency the measured reverberation time values overlap between the empty room situation and the situation with material installed into the room. Above the Schroeder frequency, the measured reverberation times have a smaller variation.

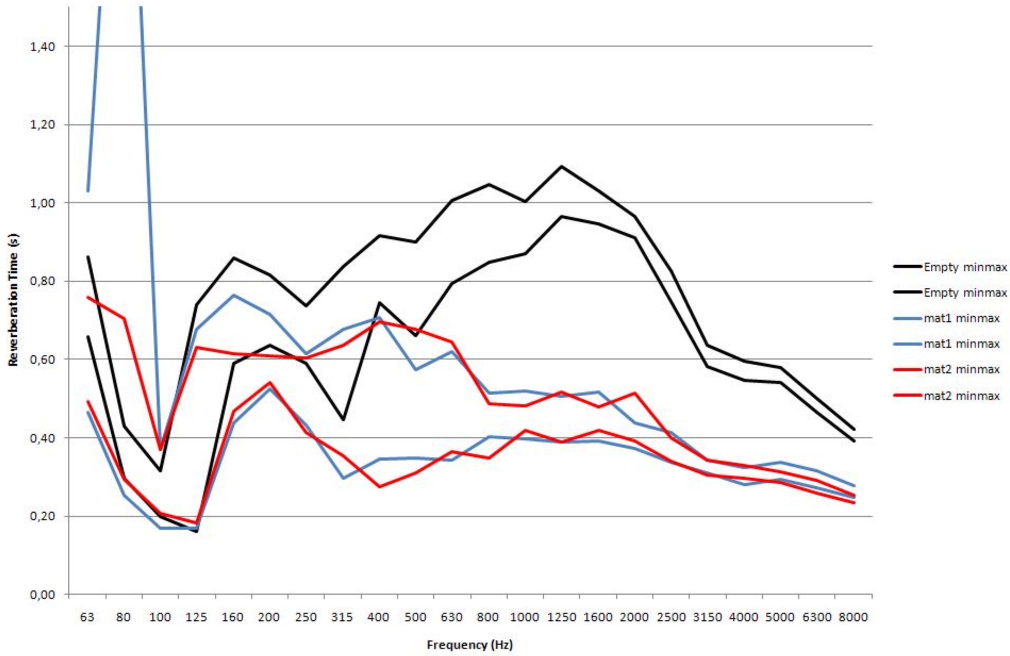

Figure 22: Minimums and maximums of measured reverberation times in the scale model

Acoustic Absorption

The acoustic absorption calculated from measured reverberation times inside the scale model can be seen in Figure 23. The variation of measured reverberation times make the results very unreliable under the scale model's Schroeder frequency. At higher frequencies the absorption values are over 1, which is possible because the calculations were done with Sabine's formula.

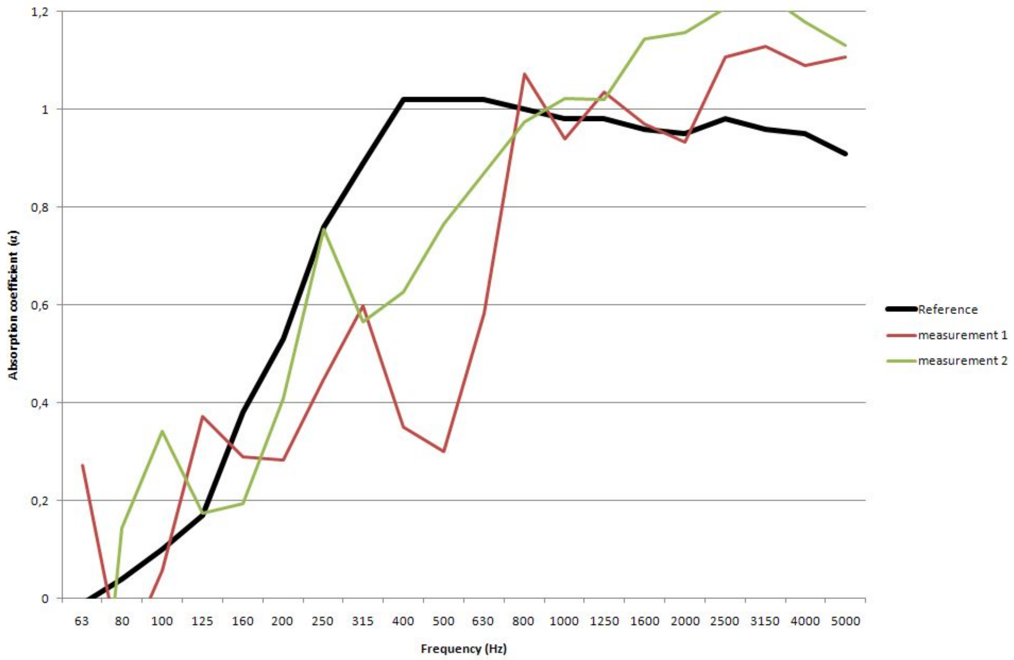

Figure 23: Acoustic absorption of the material measured in the scale model

5.2.3 Impedance Tube

The impedance tube used in the measurements was a B&K 4206. The sound source was a 3.2 inch loudspeaker which was installed inside the tube. The tube was equipped with a small detachable measurement tube for measuring at high frequencies. The tube had five couplers for mounting the microphones, three on the large tube and two on the small. The tube's technical documentation states, that the effective frequency range for the large tube should be from 50 Hz to 1.6 kHz and for the small tube 500 Hz to 6.4 kHz [15].

The equations given in Chapter 3 can be used to calculate more specific frequency limits. Table 4 show the dimensions of the tube. L denotes the large tube, S the small tube and m the microphone couplers in relation to the sample holder.

Table 4: Dimensions of the B&K 4206 Impedance Tube

d_L	100 mm
d_S	29 mm
m_{L_1}	200 mm
m_{L_2}	150 mm
m_{L_3}	100 mm
m_{S_1}	55 mm
m_{S_2}	35 mm

The low frequency limit was determined from the distance between the microphones using Equation 48, the limits were $f_{l_{13}} = 171.6$ Hz, $f_{l_{23}} = 343.2$ Hz, $f_{l_{45}} = 858.0$ Hz.

The upper frequency limit was calculated using Equation 46. For the large tube this limit was $f_{u_L} = 1990.6$ Hz and for the small tube, $f_{u_S} = 6864.0$ Hz.

The upper frequency limit also depends on the distance between the microphones. If the result of Equation 47 is lower than the result of Equation 46, the limit must be calculated according to Equation 47. Using this equation, the limit calculated between microphone couplers was 1544.4 Hz. Therefore, when microphone couplers 1 and 3 were in use, the upper limit was defined by Equation 47. However, the limit between couplers 2 and 3 was defined by Equation 46. The upper limit for the small tube was also defined by Equation 46.

In order for the impedance tube method to succeed, very precise work is required: The sample pieces must be accurately fitted into the sample holder. The sample pieces in this case were hand-made using various knives and sandpaper. ISO-10534-1 states that three different sample pieces for both tubes are required in order for the results to be reliable [8].

As can be seen from Figure 24, the edges of the sample were covered using an adhesive putty, which in this case was blu-tack, in order to seal the small gaps between the hand-made sample pices and the inner wall of the impedance tube as these gaps cause resonances, which disturb the measurements [28]. Naoki Kino

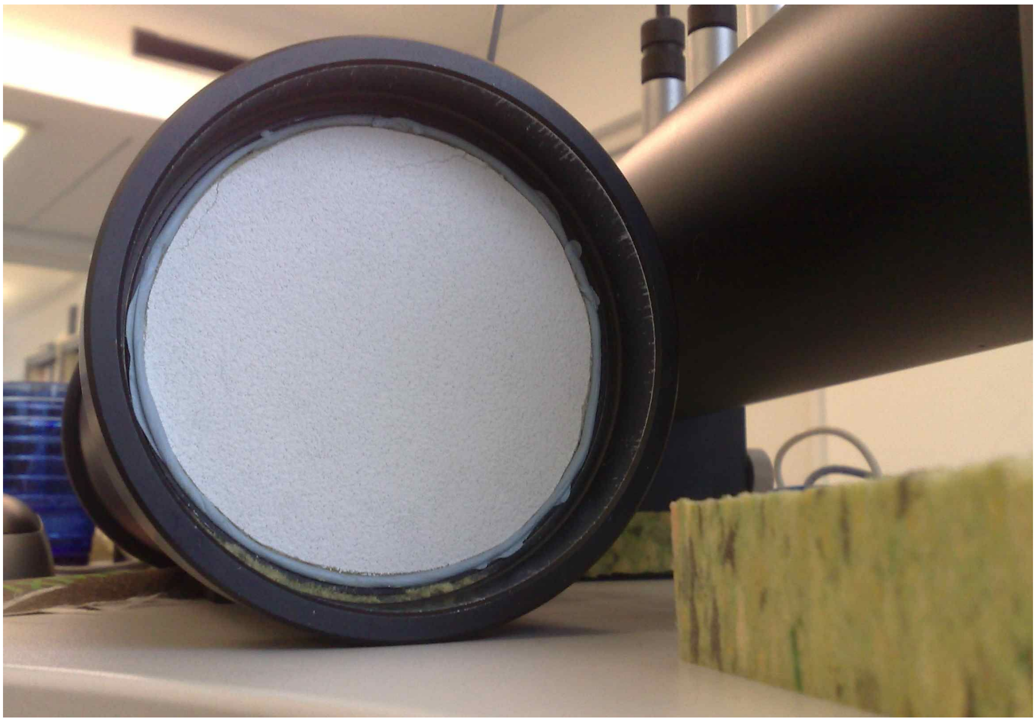

Figure 24: Impedance tube with a sample mounted

and Takayasu Ueno have concluded that the effect of these frame resonances can be lowered by choosing sample pieces that are 0.5 to 1 mm smaller than the inner diameter of the tube [14].

During the measurements it was noted that a very strong signal to noise ratio must be achieved. The analysis code written in matlab plotted very noisy absorption curves for impulse responses with SNRs between 45 and 55 dB. After switching to a different amplifier the impulse responses gave SNRs between 60 and 65 dB and the analysis started to show more clear absorption curves. The standard, however states that the SNR should be greater than 65 dB.

Acoustic Absorption

Figure 25 shows values of acoustic absorption measured for the Ecophon Master C-board using the impedance tube method. The values measured from three different sample pieces coinicide within ± 0.1 units of absorption. Where the measurement is valid for both the larger and the smaller tube the values show more variation but the trend of the absorption curve is similar with all six sample pieces.

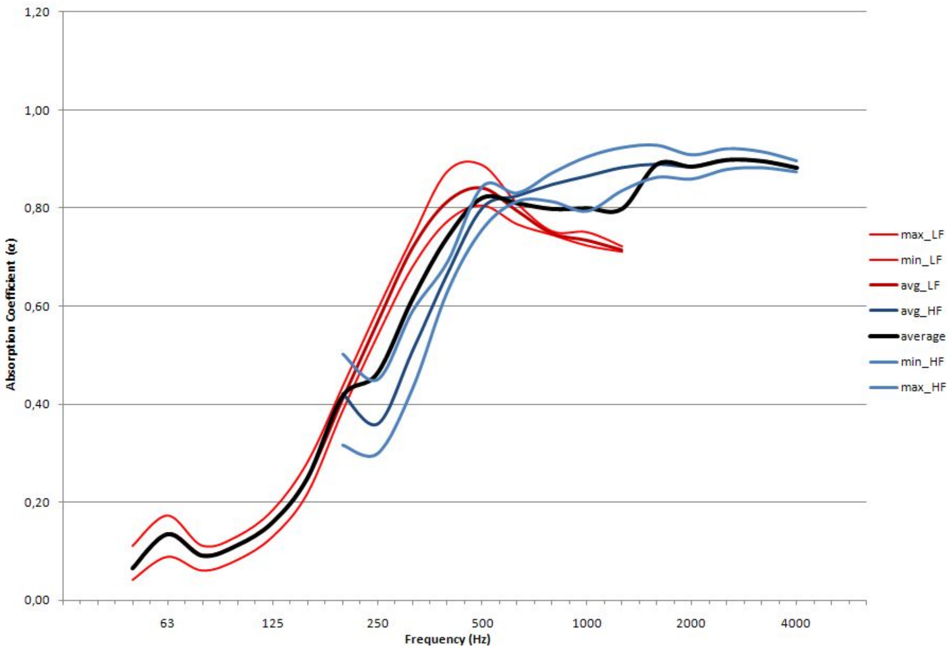

Figure 25: Values of acoustic absorption measured using the Impedance tube

6 Analysis

The purpose of this thesis was to study how various methods measure a given materials' acoustic absorption and how reliable these methods are. Laboratory results that were measured using the standardized ISO-354 reverberation chamber method were used as a reference to which the results from the alternative methods were compared to. Along with an analysis of the reliability of these methods, this chapter will also discuss other reliability issues and possible sources of errors.

6.1 Omnidirectionality of Loudspeakers

The omnidirectionality of the loudspeakers described earlier in Chapter 3 were calculated from their polar responses, according to ISO-3382 [13]. Figure 26 shows the maximum deviation of the large loudspeaker at each octave band, calculated at 30^o gliding average. The limits of deviation are plotted in the same figure [13]. Figure 27 shows the results of the same calculations for the small loudspeaker.

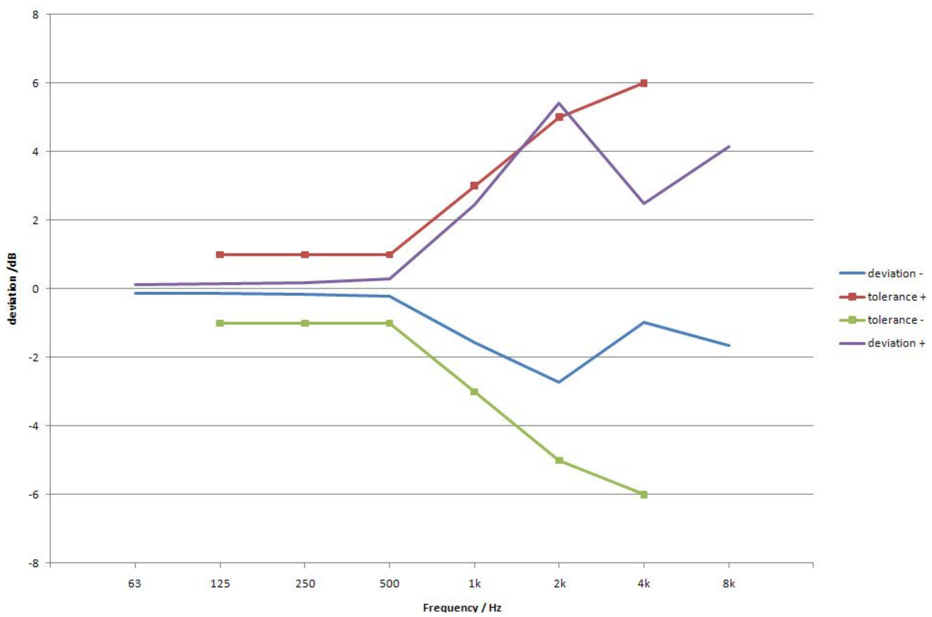

Figure 26: The omnidirectionality of the larger loudspeaker

In order to measure reverberation time reliably, two loudspeakers were measured to check if they met the omnidirectionality conditions for ISO-354 reverberation chamber measurements [3]. The loudspeakers were measured and their omnidirectionality was calculated according to ISO-3382 [13].

As can be seen from Figure 27, the small loudspeaker meets the omnidirectionality requirements according to ISO-3382 [13]. The loudspeaker is fairly omnidirectional at lower frequencies. However, frequencies above those which have a smaller wavelength than the diameter of a single element of the speaker, the individual elements start to become visible in the polar pattern. The results of the large

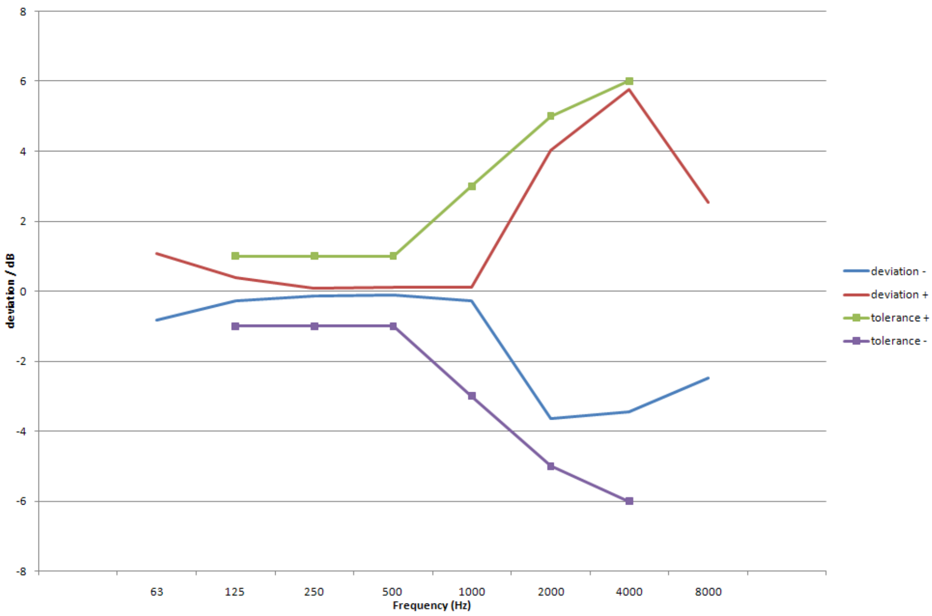

Figure 27: The omnidirectionality of the smaller loudspeaker

loudspeaker are quite similar, however at the 2000 Hz octave band the deviation becomes substantial, which makes the larger loudspeaker unsuitable for measurements according to both ISO-354 and ISO-3382 [3] [13].

The large loudspeaker was further tested to determine whether it could meet the omnidirectionality spesifications of ISO-140-4, which means that it could still be used in sound isolation measurements [2]. The calculation is similar to the omnidirectionality test described above, but the tolerance is higher [2]:

. . . "Uniform omnidirectional radiation can be assumed if the DI values are within the limits of ±2 dB in the frequency range from 100 Hz to 630 Hz. In the range from 630 Hz to 1 000 Hz, the limits increase linearly from ±2 dB to ±8 dB. They are ±8 dB for frequencies of 1 000 Hz to 5 000 Hz."

The calculations according to ISO-140-4 were done in third octave bands, which means that individual deviations might show more clearly, but the tolerance values are higher than in ISO-354 and ISO-3382 [2] [3] [13]. The results of these calculations are shown in Figure 28.

As can be seen from Figure 28, the large loudspeaker meets the requirements for sound isolation measurements, because the deviation plots do not exceed the tolerance plots [2].

6.2 Acoustic Absorption

The results of the acoustic absorption measurements for the test material are plotted along with the reference curve both in octave and third octave bands. These results can be found in Figures 29 and 30, respectively.

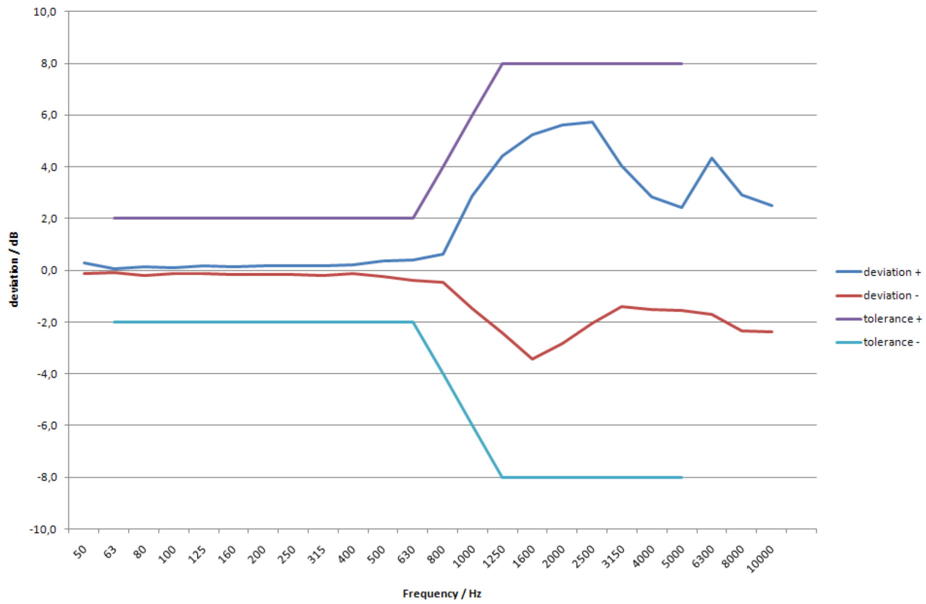

Figure 28: The omnidirectionality of the large loudspeaker

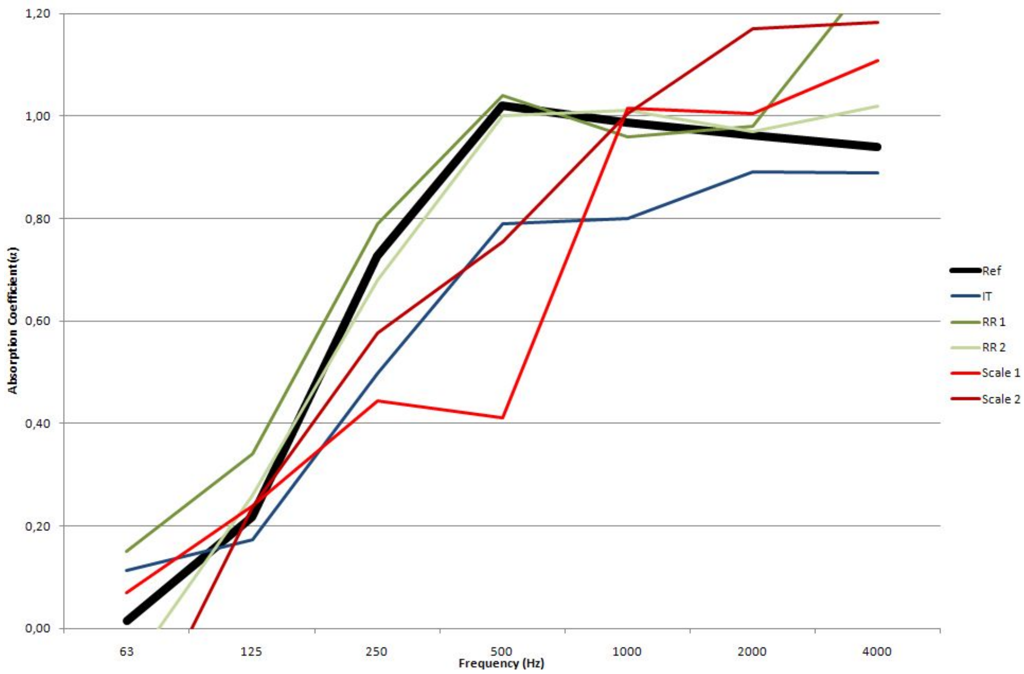

Figure 29: Octave band analysis of results provided by all methods

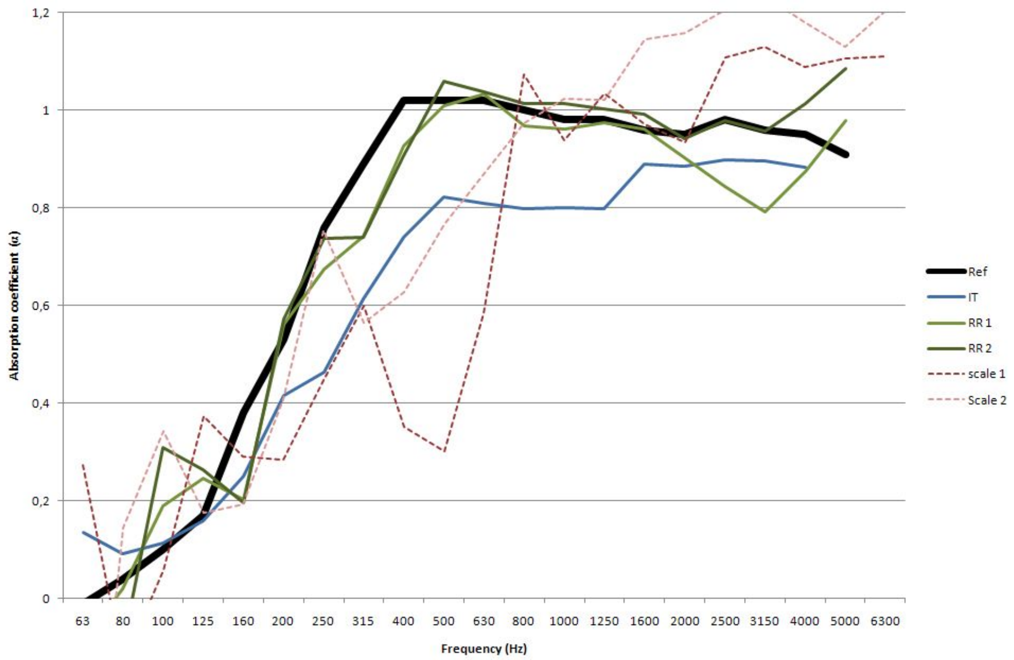

Figure 30: Third octave band analysis of results provided by all methods

6.3 Reliability

6.3.1 Reliability of Reverberation Time Measurements

An objective study of the reliability of reverberation time measurements is required, because the measured reverberation time value is represented as an average of measured impulse responses in a given space. In ISO-354 [3] an equation is given for evaluating the relative standard deviation of the measured reverberation times [3]:

$$\varepsilon_{20} = \sqrt{\frac{2.42 + 3.59/N}{fT}}, \tag{50}$$

where ε_{20} is the standard deviation of measured reverberation times over a 20 dB decay time, it therefore describes frequency dependent spatial variation in a given space. In Equation 50, T is reverberation time, f is the centre frequency of a third octave band and N is the number of measurement points inside the room. Below, in Figures 31,32, 33 and 34, the acoustic absorption of both reverberation room and scale model measurements are plotted with minima and maxima calculated according to Equation 50.

The standard for impedance tube measurements using the transfer-function method [9] states, that:

"Information concerning the reproducibility and repeatability of these test methods is not available."

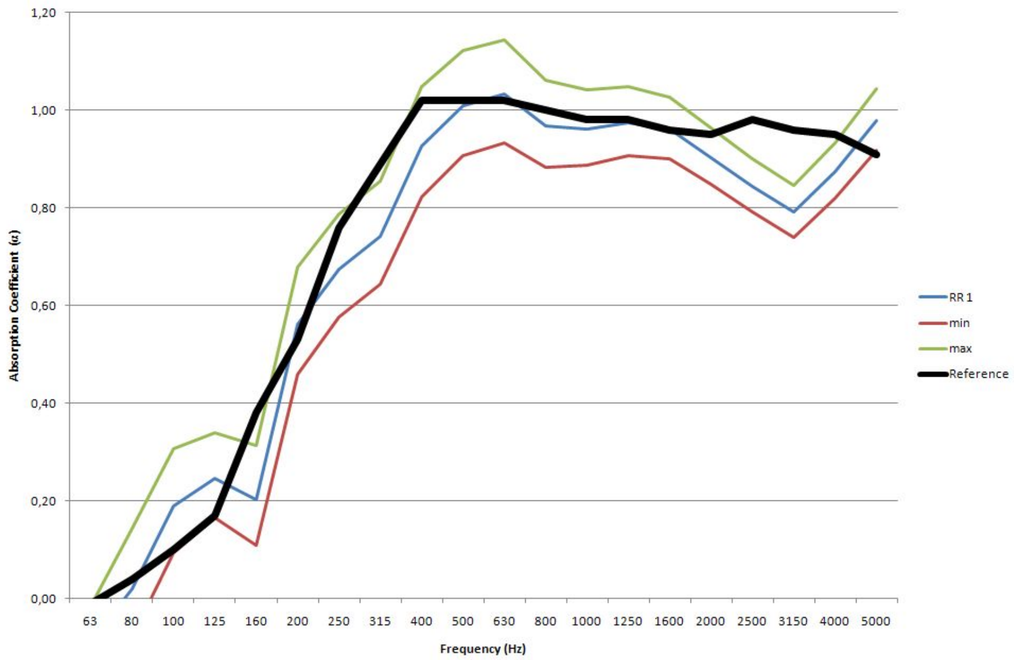

Figure 31: The acoustic absorption with standard deviation according to ISO-354 (reverberation room, first measurement)

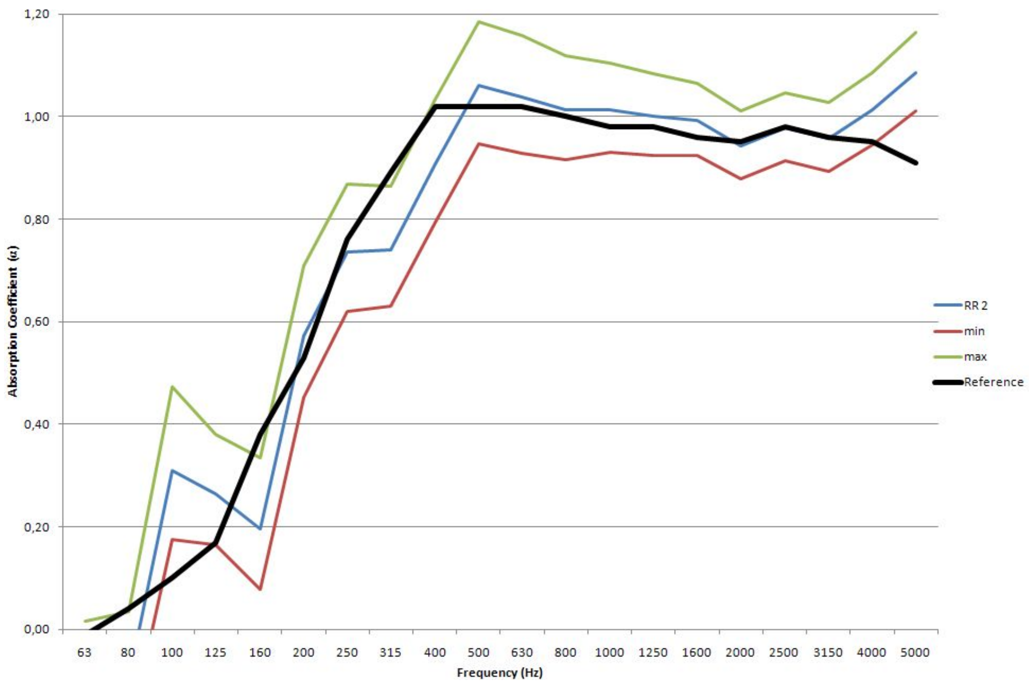

Figure 32: The acoustic absorption with standard deviation according to ISO-354 (reverberation room, second measurement)

In appendix 2, Figure 35 shows ε_{20} for the reverberation room, and Figure 36 for the scale model measurement.

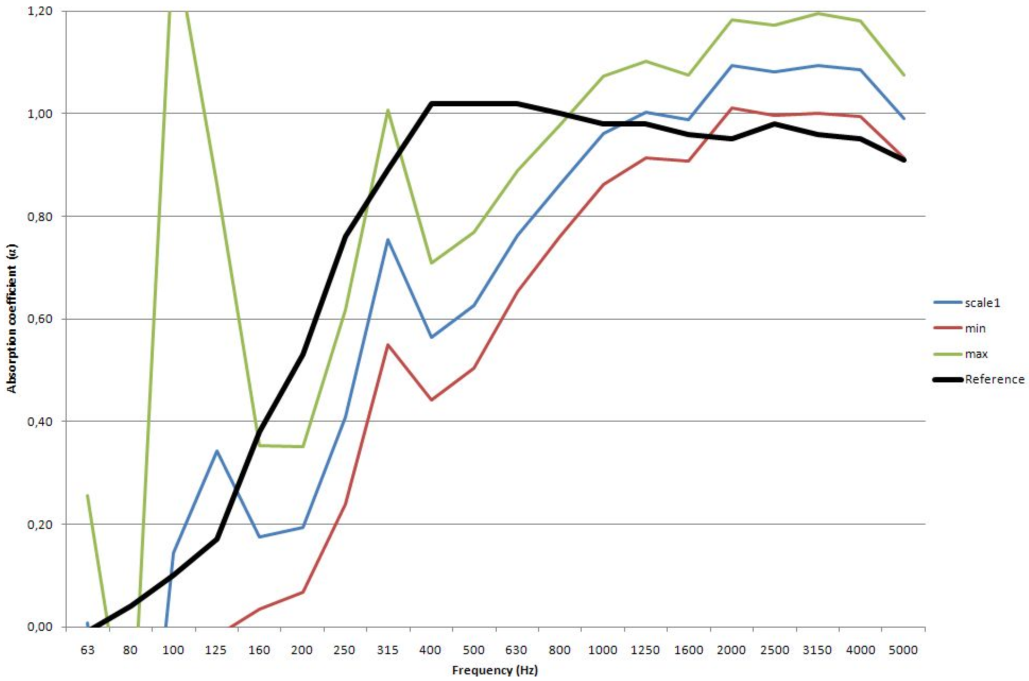

Figure 33: The acoustic absorption with standard deviation according to ISO-354 (scale mode, first measurement)

Figure 34: The acoustic absorption with standard deviation according to ISO-354 (scale model, second measurement)

6.3.2 Reliability of the Alternative Methods

In the previous section, reliability of reverberation time measurements, in general, was adressed. In this section the reliability of the alternative methods is analyzed in comparison to the reference results in order to study how well the alternative methods perform. The absorption coefficients of the material for all of the methods, at their reliable frequency ranges, are shown in Table 5, where the largest deviation in each column is indicated with a bold number.

Table 5: Reliability of all methods

f(Hz)	REF	IT	dev.%	RR1	dev.%	RR2	dev.%	S1	dev.%	S2	dev.%
63	**-0.01**	0.14	15								
80	**0.04**	0.09	5								
100	**0.1**	0.11	1								
125	**0.17**	0.16	-1								
160	**0.38**	0.25	-13								
200	**0.53**	0.42	-11								
250	**0.76**	0.46	-30								
315	**0.89**	0.61	-28	0.74	15	0.74	15				
400	**1.02**	**0.74**	**-28**	0.93	-9	0.91	11				
500	**1.02**	0.82	-20	1.01	1	1.06	-4				
630	**1.02**	0.81	-21	1.03	-1	1.04	-2				
800	**1**	0.80	-20	0.97	3	1.01	-1				
1000	**0.98**	0.80	-18	0.96	2	1.01	-3				
1250	**0.98**	0.80	-18	0.97	1	1.00	-2				
1600	**0.96**	0.89	-7	0.96	0	0.99	-3				
2000	**0.95**	0.88	-7	0.90	5	0.94	1	0.93	-2	1.16	21
2500	**0.98**	0.90	-8	0.84	14	0.98	0	1.11	13	1.21	23
3150	**0.96**	0.90	-6	**0.79**	**17**	0.96	0	1.13	17	1.24	**28**
4000	**0.95**	0.88	-7	0.88	7	1.01	-6	1.09	14	1.18	23
5000	**0.91**			0.98	-7	**1.09**	**18**	**1.11**	**20**	1.13	22

Table 5 shows that the reverberation room measurements have the lowest overall deviation from the reference values.

6.4 Error Sources

There are two main error sources in reverberation time measurements. First, any inaccuracy in the measured reverberation times will reflect as an error in absorption coefficient. As shown in Figures 19 and 22, the minima and maxima of the reverberation times, especially at low frequencies, have high deviation. The number of independent measurement points has proven not to have a significant effect on spatial variation in reverberation time accuracy, as long as the requirement of the minimum amount of measurement points described in ISO-130-13 is met [12].

Investigations on other measurable quantities than the uniform squared pressure has been found by the author and, for example, acoustic energy density measurements have been proven to provide more accurate results with fewer measurement points [21]. Energy density is spatially more uniform than traditionally measured squared pressure. The technology to measure energy density is not yet in wide use, but the excellent results should be convincing enough to make intensity measurements more widely used.

Another error source in absorption measurements, is reproducibility. Variations in results due to the use of different testing equipment and laboratory exist. It was acknowledged, that the reverberation room and scale model measurements performed for this thesis could have been repeated but, as the standard has not yet to be updated on the reproducibility issue, such testing would have been impossible to analyze.

For the impedance tube, errors are mostly due to variations in sample construction. The measurements in this thesis used the variant of the method in which all impulse responses inside the tube were measured using only one michrophone. The calculated absorption values confirm that at the frequency bands where both impedance tubes are valid, the measurement results coincide quite well. Also the SNR of individual impulse responses were within the limits of the standard, which means that noise did not affect the results.

7 Conclusions and Future Work

7.1 Performance of the Alternative Methods

In this thesis, acoustic absorption was studied and measured for a glasswool board, using one standardized and two unstandardized methods. The obtained results were compared to reference values that were provided by the manufacturers of the board. The reference values were measured according to ISO-354 [3].

The methods that were used in the measurements were the standardized impedance tube method and the two non-standardized methods were; the reverberation room method and the scale model method.

The results presented in the previous chapter show that the reverberation room method outperforms the two other methods in terms of reliability and correlation with the reference results. Even at the frequency band where the method was concluded not to be reliable, the measured values of the absorption coefficient correlate well with the reference values. At higher frequencies, it seems that both the edge effect and the gaps between the boards seem to produce very high values of absorption. This is due to the fact that acoustic absorption was calculated according to ISO-354 [3]. The standard states that absorption should be calculated using Sabines equation, which does not account for the edge effect or the gaps. On low frequencies, the effect of room modes show as a local maximum and miminum at 100 and 150 Hz. Figure 18 shows theoretical modal densities at approximately these frequencies.

The accuracy of the reverberation room measurement could have been improved, especially at high frequencies, by framing the sample or mounting it into the floor (as is done in actual reverberation chambers). When installing the sample pieces to the floor, it can be concluded that the reverberation room measurement is accurate with approximately 5 % error at a frequency band of 200 to 3.5 kHz in a room of volume 73.6 m^2, in comparison to existing results. The spatial variation of measured T_{60} values is shown in Appendix B.

The scale model measurement did not perform as well as the reverberation room method. Two similar measurements were performed using the same test sample and the results did not correlate well at the methods' theoretically reliable frequency range. The spatial variation of measured T_{60} values inside the scale model are shown in Appendix 2. When performing reverberation time measurements inside a scale model, the absolute difference between the situations are quite small, in this case approximately 0.2 seconds in the reliable range. In Figure 22, reverberation times measured with the material sample installed has small variation in the methods' reliable frequency range.

In conclusion, the reverberation room method is the only method that produces reliable enough results to allow for comparison of the room acoustic behaviour of different materials.

7.2 Other Conclusions

The impedance tube method produced reliable results at the whole frequency band specified in ISO-266 [1]. The calculated absorption coefficients were significantly lower than the reference values at mid frequencies. This result reveals more about the properties of the material than about the method.

Cube shaped loudspeakers can not exclusively be concluded to be omnidirectional, although if designed correctly, it is possible to create a cube shaped omnidirectional loudspeaker.

7.3 Reverberation Time Measurements

It is a known fact that reproducibility between laboratories is not good. During the time this thesis was written, the ISO-354 standard [3] for reverberation chamber measurements stated that:

"The reproducibility of absorption coefficient measurement is still under investigation"

A number of factors cause measurement uncertainties in qualified laboratories. It would be beneficial for both professionals and academic researchers to find methods to improve the method by minimizing the known sources of error.

Published in 2010, an article [27] states that a work group for ISO has started to investigate possibilities to improve the reverberation chamber method. A good finding of the aforementioned study for reducing inter laboratory deviations is the use of a reference absorber. A reference absorber would be sent to each laboratory and used to calibrate each reverberation chamber. The same study proved that the use of a reference absorber would increase the reproducibility to an acceptable level.

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Appendixes

A Measurement Equipment List

Table 6 contains the list of equipment used in the loudspeaker test. Table 7 contains the list of equipment used in the scale model measurement.

Table 6: Loudspeaker test equipment

Computer	Fujitsu Siemens Amilo Pa 2510
Operating system	Windows XP SP3
Software	ARTA 1.5.0
Sound Card	Marian Ucon CX
Amplifier	Crest Audio CA 6
Loudspeakers	as described in chapter 5
Michrophone	DPA 4006
Thermal and humidity measurer	Not Used

Table 7: Scale model measurement equipment

Computer	Fujitsu Siemens Amilo Pa 2510
Operating system	Windows XP SP3
Software	ARTA 1.5.0
Sound Card	Marian Ucon CX
Amplifier	Crest Audio CA 6
Michrophone	G.R.A.S 40 Bf
Michrophone amplifier	G.R.A.S Type 26 AS
Michrophone preamplifier	G.R.A.S Power module Type 12AA
Thermal and humidity measurer	Tinytag TV-4505

Table 8 contains the list of equipment used in the reverberation room measurement. Table 9 contains the list of equipment used in the impedance tube measurement

Table 8: Reverberation room measurement equipment

Computer	Fujitsu Siemens Amilo Pa 2510
Operating system	Windows XP SP3
Software	ARTA 1.5.0
Sound Card	Marian Ucon CX
Amplifier	Crest Audio CA 6
Michrophone	DPA 4006
Thermal and humidity measurer	Tinytag TV-4505

Table 9: Impedance tube measurement equipment

Computer	Fujitsu Siemens Amilo Pa 2510
Operating system	Windows XP SP3
Software	ARTA 1.5.0
Sound Card	Marian Ucon CX
Amplifier	Cambridge Audio Azur 340A
Michrophone	B & K 4187
Michrophone amplifier	B & K 2670
Michrophone preamplifier	G.R.A.S Power module Type 12AA
Thermal and humidity measurer	Not Used
Impedance tube	B&K 4206

B Reliability Studies

In this Appendix, values of ε_{20} are plotted at third octave bands. In Figure 35 it can be seen, that the spatial variation of measured reverberation time is below 0.1 seconds within the methods reliable frequency range. As for the scale model method, Figure 36 shows values below 0.02 seconds for the methods reliable frequency range.

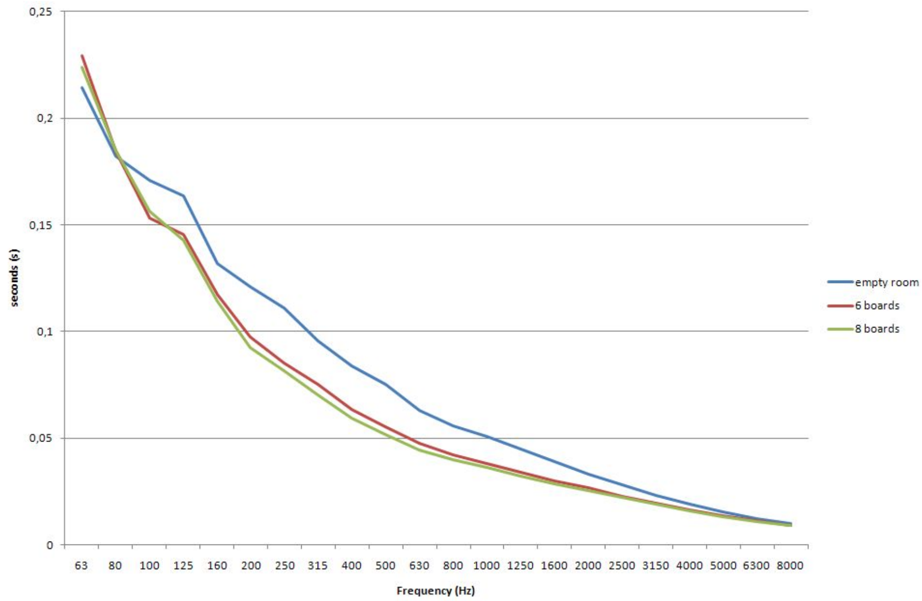

Figure 35: Third octave band plot of ε_{20} for the reverberation room measurements

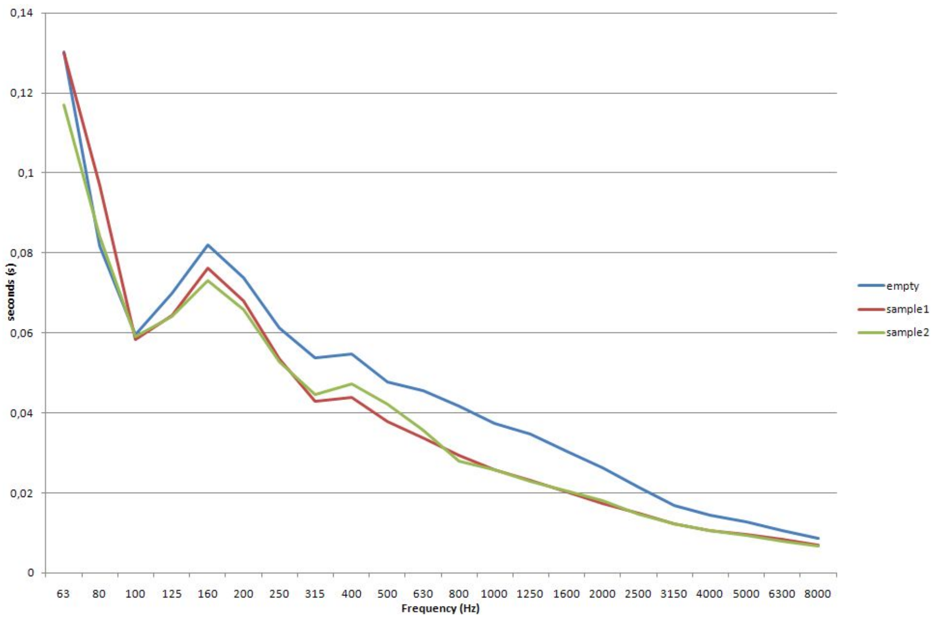

Figure 36: Third octave band plot of ε_{20} for the scale model measurements